MiVoice Office 250

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MiVoice Office 250 Features and Programming Guide
Release 6.1
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Chapter 1

NEW FEATURES

INTRODUCTION

This chapter lists new features related to the MiVoice Office 250 Database (DB) Programming application. This includes features used to complete system configuration and perform system adds, moves, and changes *after the system is installed*. For information about new features related to system hardware, licensing, or upgrades, refer to the *MiVoice Office 250 Installation Manual*.

SYSTEM DOCUMENTATION RESOURCES

MiVoice Office 250 documentation includes the following resources:

- *Mitel MiVoice Office 250 Features and Programming Guide* (this book)

- *MiVoice Office 250 Installation Manual*

- Mitel MiVoice Office 250 DB Programming Help

- *Mitel MiVoice Office 250 Phone Administrator Guide*

- *Mitel MiVoice Office 250 Unified Voice Messaging Administrator Guide*

- *Mitel System Administration & Diagnostics Guide,*

- Mitel System Administration & Diagnostics Help

For additional system documentation, refer to the "Documentation" folder of the system software DVD. You can also find all documentation on the Mitel eDocs Web site (http://edocs.mitel.com).

MIVOICE OFFICE 250 V6.1 NEW FEATURES

This section briefly describes the new features, hardware, and enhancements available on the MiVoice Office 250 when equipped with the v6.1 software release.

PRODUCT RE-BRANDING

Mitel has introduced new product naming as follows:

PREVIOUS NAME	CURRENT NAME
Mitel 5000/5000 CP/Mitel 5000 Communications Platform	MiVoice Office 250
Mitel Database Programming	MiVoice Office 250 Database Programming.
Mitel System Administration and Diagnostics	MiVoice Office 250 System Administration and Diagnostics
Mitel 5000 Administrator Web Portal	MiVoice Office 250 Administrator Web Portal
Mitel 5000 User Web Portal	MiVoice Office 250 User Web Portal
Mitel Attendant Console	MiVoice Office 250 Attendant Console
Mitel Applications Suite (MAS)	MiCollab
Mitel Unified Communicator Advanced (UCA)	MiCollab Client
Mitel Border Gateway	MiVoice Border Gateway
NuPoint Unified Messaging	MiCollab Unified Messaging

NOTE	For products related to releases prior to 6.1, naming will reference their previous, non re-branded names.

The re-branded interfaces for Database Programming, System Administration and Diagnostics, Administrator Web Portal, and the User Web Portal are updated with the new Mitel logo and color scheme. Additionally, the re-branding includes, but is not limited to, the following other MiVoice Office 250 interfaces:

• MiVoice Office 250 Installation Wizards

• MiVoice Office 250 Private Networking

• MiVoice Office 250 PS1 Server

VIRTUALIZED PS1

The PS1 can now be deployed as a virtual appliance running in a VMware environment. Please see the MiVoice Office 250 Installation Manual for details regarding the installation and upgrade to virtualized PS1.

Note: Throughout this document, PS-1 refers to either the physical dedicated hardware component or the virtual appliance, unless specified.

NEW PS1 HARDWARE SERVER

As of May 2014, Dell retired the R210 Server and replaced it with the PowerEdge R220 Server.

The R220 Server supports the MiVoice Office 250 PS1 software, and it is capable of running the 32-bit Debian 4.0 based distribution with the MiVoice Office 250 components.

NOTE	Debian is a Linux distribution that is composed primarily of free and open-source software, most of which is under the GNU General Public License, and packaged by a group of individuals known as the Debian project. The Debian Stable distribution is one of the most popular for personal computers and network servers, and has been used as a base for several other Linux distributions.

MIVOICE OFFICE 250 ATTENDANT CONSOLE REFRESH

Due to its historical heritage, earlier versions of the Attendant Console supported older operating systems such as Windows 98 and old drivers for HASP keys. These drivers do not support 64-bit versions of the Microsoft Windows.

As of 2014 new drivers for the Sentinel HASP are available from SafeNet Inc., and are compatible with a variety of 64-bit Windows versions, including Windows 7 and Windows 8.

The MiVoice Office 250 Attendant Console Release 3.5 can read data from valid HASP keys on the following Windows Operating systems:

- Windows 7 (32- and 64-bit)

- Windows 8 (32- and 64-bit)

- Windows 8.1 (32- and 64-bit)

- Windows 10 (32- and 64-bit)

DIGITAL TELEPHONE REFRESH

The Digital Telephone refresh redesigns the existing 8568 and 8528 Digital Telephone hardware to eliminate the potential for obsolete parts with the new redesigned Digital Telephones.

No new features are added to the refreshed series of Digital Telephones. To the end-user there are no differences in the installation and operation between the refreshed Digital Telephones and the previous versions.

The refreshed MiVoice 8568 and 8528 Digital Telephones are supported on all Mitel 5000 CP / MiVoice Office 250 systems that currently support these phones.

VIDEO SUPPORT FOR SIP TO SIP CALLS

MiVoice Office 250 Release 6.1 supports video capable SIP endpoints connected to the system when one SIP endpoint tries to establish a video call or escalate an established audio call to video. The peer-to-peer (P2P) functionality enables new SIP Peer endpoints to send and receive video packets directly to another IP device.With P2P enabled, an existing SIP to SIP audio call will have the capability to be converted to a video call.

In Database Programming, *Peer-to-Peer Audio* has been renamed *Peer-To-Peer Media*.

BT CLIP CLID ENHANCEMENTS

MiVoice Office 250 Release 6.1 allows the decoding of Calling Line Identification (CLID) information in countries which use variations of BT CLIP (British Telecom Calling Line Identification Presentation) signaling on analog trunks.

BT CLIP supports the delivery of Caller ID information on analog trunks in the UK, and the standard is commonly known as BT CLIP. BT CLIP is an on-hook capability that provides the user with information about the caller before actually answering a call.

A new trunk Service Type and timer parameters are added in MiVoice Office 250 Database Programming to support BT CLIP. BT CLIP replaces ETSI DTMF Caller ID. When configuring a Loop Start trunk, the Service Type options now include Bellcore FSK Caller ID and British Telecom FSK.

See "Caller ID, DNIS, and ANI" on **page 172** for details.

SIP ACD AGENT LOGIN

SIP endpoints - including a SIP DECT, SIP Softphone, and a Mitel Phone Manager Softphone to login in to and logout from an ACD queue is supported.

SIP phones can be added to a Basic or UCD Hunt Group member list. Calls routed to a particular Hunt Group may be directed to a SIP phone.

For a SIP phone to use the ACD features (Log in, Log out, Wrap-Up Terminate), an OAI application must be used in conjunction with the SIP phone. Mitel Phone Manager is the recommended application.

The ACD feature codes cannot be dialed using a SIP phone.

See **Table 113** and **Table 114** for a full list of system features and phone features compatible with each type of SIP phone.

5000 CP V6.0 SP3 NEW FEATURES

This section briefly describes the new features, hardware, and enhancements available on the Mitel 5000 CP when equipped with the v6.0 SP3 software release.

There are no impacts to any of the 5000 CP documents or help files as a result of the following new features in release 6.0 SP3.

SHELLSHOCK SECURITY UPDATES

Shellshock is a family of security bugs in the widely used Unix Bash shell. The Shellshock vulnerabilities affect various Unix-based systems use to execute command lines and command scripts.

The 5000 CP v6.0 SP3 has included a series of fixes to prevent the Shellshock vulnerabilities from affecting the system.

NEW 6.2.0.6 IP PHONE FIRMWARE

5000 CP v6.0 SP3 introduces a new IP firmware version 6.2.0.6. The 53xx series IP Phones will be upgraded to this version once the 5000 CP v6.0 SP3 is installed.

MITEL SIP DECT 6.0 SUPPORT

In 5000 CP v6.0 SP3, the SIP DECT Version 6 offering is fully supported and validated.

MICROSOFT EXCHANGE 2013 SUPPORT FOR UNIFIED VOICE MAIL

In 5000 CP v6.0 SP3, Microsoft Exchange 2013 is fully supported when using it in conjunction with the Unified Voice Mail integration.

5000 CP V6.0 SP2 NEW FEATURES

This section briefly describes the new features, hardware, and enhancements available on the Mitel 5000 CP when equipped with the v6.0 SP2 software release. Some of the items are documented mainly in this manual, and some in their respective application's documentation set. For details on each item, refer to the indicated pages, document, or help file.

DTMF DECODING PAYLOAD

SIP Peers in Database Programming can be configured with the necessary custom payload type value, depending on the supported payload type of the far end SIP provider. In previous releases, this value was configured in IP Settings. In Release 6.0 SP2 and up, the DTMF Decoding Payload is now located in the individual configurations for SIP Phone Groups, SIP Trunk Groups, and SIP Voice Mails. The payload type is typically negotiated between carriers and devices, but in some cases the SIP service provider does not accept requests. The DTMF Payload Type is made available by the SIP trunk service provider.

Database Programming now allows the DTMF Payload type to be selected for SIP Peers, from a range of values that correspond to those supported by the SIP service provider.

The default DTMF Decoding Payload value applies to North American, UK, and Australian systems.

See "Configuring SIP Peer Programming Options" on **page 717**, and "DTMF Decoding Payload" on **page 727** for details.

PROPAGATION OF ORIGINAL CALLER ID TO P-ASSERTED IDENTITY

In Database Programming, a new flag, *Do Not Propagate Original Caller ID to P-Asserted Identity*, was added. This flag allows the administrator to control whether or not the Original Caller ID will be propagated to the P-Asserted Identity header for SIP calls.

See "Propagate Original Caller ID" on **page 523**, and "Do Not Propagate Original Caller ID to P-Asserted Identity" on **page 524** for details.

DO NOT USE OUT-BOUND USERNAME IN REGISTER

In Database Programming, a new flag, *Do not use Out-bound Username in REGISTER*, was added. This flag controls the registration of a SIP call when upgrading to Release 6.0 SP2.

If the flag is set to No, the to/from/contact headers in the SIP register correspond to the out-bound username. If the flag is set to Yes, the to/from/contact headers in SIP register correspond to the trunk group Caller ID.

See "Authentication" on **page 719** for details.

SUPPORT FOR NEW DDM16 AND DEM16 CIRCUIT BOARD REVISIONS

New hardware revisions of the DDM16 and DEM16 circuit boards are fully supported by the Release 6.0 SP2 software.

MCD INTEROPERABILITY SUPPORT

The Mitel 5000 Release 6.0 SP2 maintains all current support with the published MCD Interoperability.

MICROSOFT LYNC INTEGRATION SUPPORT

The Mitel 5000 Release 6.0 SP2 now has support for Microsoft Lync via the Micollab 6.0 product. Refer to the MiCollab 6.0 documentation for details on this integration.

SUPPORT FOR NEW DEVICES

The Mitel 5000 Release 6.0 SP2 supports the following new devices:

- The 5610 IP Dect Phone is now supported via a Media Border Gateway connection to the Mitel 5000 CP.

- New models of the 5320e phone have a back-lit display. The Mitel 5000 v6.0 SP2 contains new firmware for the 5320e IP phones.

- Apple IOS support via a Media Border Gateway connection to the Mitel 5000 CP. Refer to the Media Border Gateway and IOS documentation for details.

- Mitel Phone Manager (formerly by Xarios).

UPS MONITORING SUPPORT CHANGES ON 6.0 SP2 AND LATER SOFTWARE

Effective Release 6.0 SP2, monitoring capabilities for previously approved suppliers of UPS units cannot be guaranteed to be compatible with the UPS Monitoring feature. See the *Mitel 5000 CP Installation Manual* for details.

5000 CP V6.0 SP1 NEW FEATURES

This section briefly describes the new features, hardware, and enhancements available on the Mitel 5000 CP when equipped with the v6.0 SP1 software release. Some of the items are documented mainly in this manual, and some in their respective application's documentation set. For details on each item, refer to the indicated pages, document, or help file.

MID CALL FEATURES

Mid-Call Features provide a way for a mobile user to perform hold, transfer, and conferencing features when the 5000 places a call via Dynamic Extension Express (DEE).

> **NOTE**   Mid-Call Features are only supported with other networked 5000 CPs running Release v6.0 SP1 and higher software.

See "Audio for Calls Holding for this Device" on **page 517**

See "User Programming Fields" on **page 788**

See "User-Related Information" on **page 806**

See "Mid-Call Features" on **page 828**

MBG SIP SERVICES

The MBG (Mitel Border Gateway) is a platform for the secure deployment of multiple network connectivity services in a number of network edge scenarios. New MGB SIP features include:

- SIP device support
  See "MBG SIP Services – Trunks and Devices" on **page 705**

- UCA application support
  See "Configuring MiVoice Office 250/UCA with MBG" on **page 707**

ADDING LOCAL DIALING PREFIXES

For Incoming calls on SIP trunks in the UK and Australian markets, where the leading prefix was not presented, the Mitel 5000 will now automatically add the following dialing prefixes:

- 0 for national calls in the UK and Australian markets

- 00 for international calls in the UK market

- 0011 for international calls in the Australian market

This will allow the called party to redial the external number back successfully, and it will allow incoming calls to match System Speed Dial numbers.

DATABASE PROGRAMMING UPDATES

The following updates to Database Programming have been implemented to improve usability as more routines become automated requiring less input from the administrator:

- Configuration Wizard suggesting Voice Mail applications types.

- Configuration Wizard Summary page lists Voice Mail applications programmed from within the Wizard, and those that have not been programmed.

- Phone users have the ability to access the Administrative Web Session via User Login/ Password.

- Create an unassociated user.

- After upgrading to Release 6.0 SP1, only the System Administration and Diagnostics (Sys-Ad) shortcut displays in the Windows Start menu.

MITEL 5000 CP ENGINEERING GUIDELINES

Formerly an appendix to Mitel 5000 CP Installation Manual, the engineering guidelines for the 5000 CP have been created in a stand-alone document. See the *Mitel 5000 CP Engineering Guidelines* for more information.

5000 CP V6.0 NEW FEATURES

This section briefly describes the new features, hardware, and enhancements available on the Mitel 5000 CP when equipped with the v6.0 software release. Some of the items are documented mainly in this manual, and some in their respective application's documentation set. For details on each item, refer to the indicated pages, document, or help file.

AXXESS NETWORKING SUPPORT

Further to product bulletins released during the 5.1 release cycle, Axxess networking is no longer supported beginning with the v6.0 release of the 5000 CP.

CONFIGURATION WIZARD

The Configuration Wizard has been updated to make start-up easier, and to easily configure a variety of system level parameters including IP and e-mail settings, expansion modules, IP phones, voice processor options, and a variety of devices.

See "Launching the Configuration Wizard" on **page 59**.

USER CREATION WIZARD

The User Creation Wizard allows an administrator to create and edit individual users and their corresponding features and extensions from one area.

See "Creating Users" on **page 790**

DATABASE PROGRAMMING ENHANCEMENTS

Database Programming has been enhanced in a number of areas. The items below reflect the areas affected.

"Maintenance Accounts" on **page 48**

"Associated Extensions" on **page 392**

"Programming Trunk Group Options" on **page 506**

"Audio for Calls Holding for this Device" on **page 517**

"Configuring SIP Peer Programming Options" on **page 717**, including the following:

- "Use Registered Username" on **page 726**
- "Disable Domain Validation" on **page 727**
- "Supports Display Updates" on **page 727**
- "Supports Ad Hoc Conferencing" on **page 727**

"User Programming Fields" on **page 788**

"User-Related Information" on **page 806**

"Dynamic Extension Express Examples" on **page 810**

"Ring Duration" on **page 813**

"Programming Dynamic Extension Express" on **page 816**

"Area Flags" on **page 125**

"Importing User Entries from CSV Files" on **page 796**

"Device Baseline Extensions" on **page 132**

DHCP SERVER

The 5000 CP now has an embedded DHCP server. The embedded DHCP server makes the 5000 CP capable to automatically provide the proper DHCP Options for programming IP phones, thereby reducing the chances of misconfiguration, and reducing installation time.

See "DHCP Server Settings" on **page 612** for details.

BUNDLED SAVE AND RESTORE

The Database Operations menu is renamed Backup Operations. Two new options for database backup save and restore are located here. The Default Database option is now under the main Operations menu.

The ability to encrypt sensitive data is also provided.

The DB Test and Repair and DB Converter user interfaces will be redesigned and incorporated into the new Backup Utility window. Scheduled backups will be updated to allow for a bundled save.

See "Backup Operations" on **page 60** for details.

CONNECTION WIZARD

If the Configuration Wizard is not displayed at startup, a new dialog listing connection tips for the Processing Server to connect with the Base Server will display instead of the Connection Wizard. This dialog will only be displayed when connecting to a Processing Server in online mode that is disconnected from the Base Server.

See "The Connection Wizard" on **page 21** for details.

SIP PEER TO PEER

A SIP Peer can now be put into a Network Group that supports peer-to-peer audio. The Peer-to-Peer (P2P) functionality enables the new SIP Peer devices to send and receive audio packets directly to the other IP device (without backplane connections). This implementation is the same as the existing 5000 IP devices.

See "SIP Peer-to-Peer" on **page 590** for details.

SUPPORT FOR ETSI DTMF CALLER ID

Caller ID information is decoded in countries which use varieties of ETSI DTMF signaling for caller information.

When configuring a Loop Start trunk, the Service Type options now include Bellcore FSK Caller ID and ETSI DTMF Caller ID.

See "Caller ID, DNIS, and ANI" on **page 172** for details.

ADMINISTRATOR WEB PORTAL

The Administrator Web Portal (AWP) interface for the 5000 CP replaces the Administrative Web Session (AWS) as of Release 6.0. The AWP maintains the functionality of the AWS, and also allows simple adds, moves, and changes for users and phones. See the AWP application for details.

MITEL BORDER GATEWAY (MBG) SUPPORT

The following 53xx series IP phones for Teleworker operation are now supported when an MBG is connected to a Mitel 5000 CP:

- 5304 (supported only with firmware 2.1.1.2 or later)
- 5312
- 5320/5320e
- 5324
- 5330/5330e
- 5340/5340e
- 5360

Depending on whether an MBG is connected to the 5000 CP network, and remote 53xx series phones are outside the network, the appropriate NAT Address Type setting needs to be programmed. See "NAT Address Type" on **page 645**.

SUPPORT FOR NEW SIP PHONES

Mitel 5603/5604/5607 SIP phones are now supported. Display updates and Ad-Hoc Conferencing using these new SIP phones is also supported.

See "SIP Phones and SIP Phone Groups" on **page 673**.

ALARMS/ALERTS VIA E-MAIL

This new feature in SysAd adds the ability to program user notification for alarms. Users are created and managed in DB Programming, as is their notification destination (e-mail, text message, etc.). In SysAd the alarms can be enabled or disabled of notification on a per user and individual alarm and/or alarm group basis.

INACTIVITY TIMER

System Administration & Diagnostics and Database Programming now use the same timer for inactivity. The timer is programmed in the Options window on the Advanced Tab in System Administration & Diagnostics.

MITEL UC 360 COLLABORATION POINT SUPPORT

The 5000 CP supports the Mitel UC360 Collaboration Point, and it is programmed as a SIP device. Details are included in the following sections:

"Viewing System Phones" on **page 354**.

"SIP Phones and SIP Phone Groups" on **page 673**.

"Maximum Number of Calls" on **page 725**.

UVM E-MAIL SYNC WITH GMAIL AND OFFICE 365

The UVM application provides integration and synchronization capabilities between a user's voice mailbox and their e-mail client application and is referred to as E-Mail Synchronization. It currently has the ability to synchronize with e-mail using IMAP with Exchange, Lotus Notes and Groupwise.

With the popularity of hosted e-mail providers, the purpose of this enhancement is to extend the capabilities to include Gmail and Office 365.

See the *MItel 5000 Communications Platform Unified Voice Messaging E-Mail Synchronization Administrator Guide* (available on eDocs at http://edocs.mitel.com/default.htm#5000_anchor) for details.

DTMF PAYLOAD TYPE 101

DTMF payload type 101 (becoming more common as the preferred standard in SIP calls) is now supported.

See General IP Settings on **page 603** for details.

TERMINOLOGY CHANGES

To accommodate changes in technology and/or to better align with the overall corporate marketing strategy, the following product terminology changes have been implemented in the v6.0 release:

- The Database Operations menu is renamed Operations

- System Administrative Accounts is renamed Maintenance Accounts

- The Administrative Web Session (AWS) is renamed the Administrative Web Portal (AWP)

Chapter 2

ABOUT DATABASE PROGRAMMING

INTRODUCTION

This guide provides descriptions and procedures for performing common administrative tasks using the MiVoice Office 250 Database (DB) Programming application. This includes instructions to complete system configuration and perform system adds, moves, and changes after installation.

The guide assumes that the system is installed and that test calls have been placed to verify that the system is properly connected to Central Office (CO) lines. It does not cover installation programming, which includes hardware programming, system upgrades, and licensing.

For more information about installation programming, refer to the following resources:

- *MiVoice Office 250 Installation Manual*

- MiVoice Office 250 DB Programming Help

    To access MiVoice Office 250 DB Programming Help:
    While in DB Programming, select Help – **Help Topics**, or press **F1**.

- *Mitel System Administration & Diagnostics Guide*

- Mitel System Administration & Diagnostics Help

CHAPTERS IN THIS GUIDE

This guide includes the following chapters:

- **Introduction**: Provides an overview of the new features and enhancements in this release.

- **About Database Programming**: Provides information about the document's structure and content, associated documents, an overview of DB Programming and programming guidelines.

- **System Management**: Describes features that you can use to manage and maintain the MiVoice Office 250.

- **Private Networking and System Nodes**: Provides information to help you configure network settings or add system nodes to a private Internet Protocol (IP) or T1/E1 Primary Rate Interface (PRI) network.

- **Numbering Plans**: Provides information to program MiVoice Office 250 Numbering Plan options.

- **Trunks and Gateways**: Describes how to program trunks and gateways for the system.

- **End User Features**: Describes MiVoice Office 250 features that are used by or mainly affect end users.

- **Phones and Devices**: Provides information about how to program phones and devices.

- **Extension Lists and System Groups**: Provides programming instructions for MiVoice Office 250 hunt groups, trunk groups, network groups, and extension lists.

- **System and Device IP Settings**: Describes MiVoice Office 250 IP features and functionality.

- **SIP Peers**: Provides programming instructions for SIP peer (SIP-enabled) phones, trunk groups, and voice mails.

- **System Settings**: Describes features that you can use to customize your system settings.

- **Users**: Provides programming instructions for creating Users in the Users folder.

- **Voice Processor Features and Programming**: Describes voice processing system features and programming for the MiVoice Office 250.

- **Subscriber Mailboxes**: Provides information to create and configure subscriber mailboxes for Unified Voice Messaging (UVM) systems.

- **Voice Processing Management**: Describes tools that you can use for UVM maintenance and how to save or restore UVM databases.

- **Voice Processor Reports**: Provides information to generate customized system voice processing reports.

- **System Diagnostics**: Provides fundamental instructions for interpreting the output data from the system diagnostic utilities.

SYSTEM ADMINISTRATION & DIAGNOSTICS

System Administration & Diagnostics is a PC-based application that provides a single interface to quickly access all of the administration and diagnostics tools available on the MiVoice Office 250. Using this application, a system administrator can easily configure and upgrade the system, monitor active calls, view resource usage, troubleshoot issues, and so forth.

Mitel MiVoice Office 250 Database (DB) Programming is installed as part of the System Administration & Diagnostics application along with other supporting system management tools and utilities. After installing the System Administration & Diagnostics application, administrators can quickly launch any of the following:

- DB Programming

- Administrative Web Portal (AWP)

- Secure Shell (SSH) Connection (PuTTY)

- MiVoice Office 250 Database Utilities (which includes Database Test and Repair)

- MiVoice Office 250 MOH Utility (the Music on Hold utility)

- Upload Utility

After establishing a connection to a MiVoice Office 250 node, System Administration & Diagnostics queries call, resource/device, system, and system status data from the node. It then uses a variety of content controls to display this information in a well-organized manner that allows administrators to easily read and understand the data. The data can be refreshed manually, or the application can be configured to refresh it automatically every so often.

The System Administration & Diagnostics installation includes an option to uninstall previous installations of DB Programming. This option appears only when previous installations are detected. Mitel highly recommends that you allow previous DB Programming versions/utilities to be uninstalled, as this will prevent any confusion as to which versions of the applications are

being used. The older versions of DB Programming can then be re-installed using the DB Programming "plug-ins" concept, as described below.

Plug-Ins

After the v4.0 release, newer versions of DB Programming will be installed as "plug-ins" so that the main System Administration & Diagnostics application does not have to be re-installed just to get a newer version of DB Programming. Note that you cannot install a plug-in release unless the full System Administration & Diagnostics release is already installed.

The correct method for deployment on a new PC, or to update a PC with the application already installed, is to install the most recent version of the System Administration & Diagnostics application (which will usually include the most recent Database Programming version), then add the required plug-ins for all previous Database Programming versions.

For complete information on this application, refer to both the *Mitel System Administration & Diagnostics Guide* and the help accessible from the application's user interface.

TECHNICAL SUPPORT

You can search the following Knowledge Base (KB) Center regarding DB Programming issues:

Mitel OnLine Knowledge Base Center (http://domino1.mitel.com/prodsupp/prodsupkb.nsf/WebSearchForm?OpenForm). Note that you must first log on to Mitel OnLine (http://portal.mitel.com/wps/myportal/MOLHome) in order to access this KB.

The KB is a current repository for resolved issues and common questions involving Mitel products. If the problem persists, contact Technical Support.

NOTICE
The following certification is required to install this equipment and to receive technical support: • MiVoice Office 250 Basic certification • Convergence Technology Professional (CTP) certification Technical support is provided for authorized products only.

PLANNING THE PROGRAMMING SESSION

Before programming, determine the features that meet your organization's needs, and then refer to the specific programs and program planning sheets. Program planning sheets and all MiVoice Office 250 Database Programming documentation can be found on the Mitel eDocs Web site (http://edocs.mitel.com).

PROGRAMMING WIZARDS

DB Programming includes several wizards that allow you to quickly program different parts of the system without having to navigate to separate areas in DB Programming. Wizard details and instructions are included in the appropriate sections.

The Configuration Wizard

The Configuration Wizard allows you to quickly configure a system with enough basic information to get the system up and running. Verify the system type to make sure it matches the system you have. If there is a difference, either the hardware is incorrect or the license type is incorrect. See "Launching the Configuration Wizard" on **page 59** for instructions on how to launch the wizard. For complete information about the Configuration Wizard, refer to the MiVoice Office 250 DB Programming Help.

The Networking Wizard

The Networking wizard allows you to quickly configure IP networking or T1/E1 PRI networking for existing or new nodes. See "Launching the Networking Wizard" on **page 99** for instructions on how to launch the wizard. For complete information about the Networking Wizard, refer to the *MiVoice Office 250 DB Programming Help*.

The Connection Wizard

The Connection Wizard is a troubleshooting tool that helps you to establish a connection to the Base Server. If the Configuration Wizard is not displayed at startup, a new dialog listing connection tips for the Processing Server to connect with the Base Server will display instead of the Connection Wizard. This dialog will only be displayed when connecting to a Processing Server in online mode that is disconnected from the Base Server.

For complete information about the Connection Wizard, refer to the *MiVoice Office 250 DB Programming Help*.

PROGRAMMING CHECKLIST

After all system hardware and software is installed, perform the programming in the order discussed here.

Table 1: Programming Checklist

TO DO	DONE
1. System management options such as passwords, database backup options, and system reset options. See "System Management" on **page 43**.	☐
2. System nodes. See "Private Networking and System Nodes" on **page 91**.	☐
3. Numbering plans for all system nodes. See "Numbering Plans" on **page 123**.	☐
a. Change the home area codes (does not apply to European systems).	☐
b. Set up ARS route groups, dial rules, and facility groups.	☐
c. Program toll restriction area flags, emergency number information, toll strings, classes of service, device baseline extensions, and user groups (area flags and user groups are not applicable to European systems).	☐
4. Trunk options for all system nodes. See "Trunks and Gateways" on **page 159**.	☐
5. Phones and devices (including phantom devices) for all system nodes. For example, program phones, off-node devices, account codes, attendants, phone flags, keymaps, message centers, Do-Not-Disturb and Reminder messages, primary attendants, system forwarding paths, and System Speed Dial. See "Phones and Devices" on **page 353** and "End User Features" on **page 273**.	☐
6. Groups and lists. Program extension lists, trunk groups, network groups, and hunt groups. See "Extension Lists and System Groups" on **page 499**.	☐
7. System and device IP settings and IP connections for system nodes. See "System and Device IP Settings" on **page 599**.	☐
8. SIP Peers (SIP Phones, SIP Phone Groups, Service Provider SIP Trunks, SIP Trunk Groups, and SIP Voice Mails) See "SIP Peers" on **page 671**.	☐
9. System settings.	☐
a. Program system page zones, system flags, and system timers and limits.	☐
b. Change any feature codes that conflict with extension numbers or that the customer wants to change. See "System Management" on **page 43**.	☐
10.User settings. See "Users" on **page 785**.	☐
11.Voice processor system programming. See "Voice Processor Features and Programming" on **page 835**.	☐
a. Create voice processor applications such as Message Notification/Retrieval (MNR), Call Routing Announcements (CRAs), Record-A-Call, Schedule Time-Based Application Routing (STAR), and so forth.	☐
b. Create a voice mail administrator mailbox.	☐
c. Program other voice processor applications and settings, extension IDs, group lists, voice processor options, timers, custom recordings, and system information.	☐
d. Program feature-specific voice processor settings such as VPIM Networking.	☐
e. Voice processor maintenance features such as Auto Reboot and Auto Scheduling options.	☐
12.Subscriber mailboxes and additional feature-specific mailbox settings such as Fax-On-Demand and E-mail Synchronization. See "Subscriber Mailboxes" on **page 959**.	☐
13.Maintenance system options. Program call cost, freeze zones, Message Print, and SMDR information, if applicable. See "System Management" on **page 43**.	☐
14.Save the system database. See "Launching the Configuration Wizard" on **page 59**.	☐

STARTING A DB PROGRAMMING SESSION

You must launch the System Administration & Diagnostics application to start a DB Programming session. Refer to the *Mitel System Administration & Diagnostics Guide* for details.

Authentication

When launching v4.0 or later DB Programming, the system asks for a username and password if you are not already authenticated through System Administration & Diagnostics. If the username or password is not valid, an error message appears, and the system asks for a username and password again. If authentication fails three times, DB Programming shuts down.

Keep the following considerations in mind:

- **Launching an offline session**: If the admin account password is set to the default password (itpassw), the system does not request authentication. Even if additional accounts have been enabled or programmed, or the database was converted from an earlier version of DB Programming, as long as the admin account password is default, you can launch an offline DB Programming session without authentication.

- **Launching an online session**: The only system account enabled in a default database is the admin account. If the admin account password is set to the default password (itpassw), the first time you launch an online session the Change Password dialog box appears, and you are forced to change the default password. The new password is automatically updated in System Administration & Diagnostics so that subsequent online sessions will not prompt for a password.

NOTE	Whenever a password is changed in DB Programming during an online session, System Administration & Diagnostics is updated to match the new password.

DB PROGRAMMING USER INTERFACE

The DB Programming user interface, Mitel DB Programming (as shown in **Figure 1**), allows you to quickly move between programming options. For more information about Mitel DB Programming elements, see **page 26**. You can see both the hierarchy of the database and all the items in each programming area. When you click an element or folder in the left pane, the contents appear in the right pane.

Beginning in Release 5.1, the DP Programming window has been updated to reflect the following design elements:

- Windows Vista style controls

NOTE	Although the DB Programming window displays using the Windows Vista style folders view by default, you can revert back to the Classic Style. Windows Vista style controls (such as dialogs and buttons) remain. Some of the graphic representations in this guide are shown using the Windows Vista style.

- Several icons updated using current standards, modern styling and standard Microsoft images wherever possible and appropriate

- Tool bar buttons updated:

  - As noted above

  - The **Refresh** button has been removed but this option is still available in the View menu

  - The **Collapse All** button has been removed but this option is still available in the View menu

  - The **Help** button has been removed but this option is still available in the Help menu

- Navigation bars replaced with Windows 7 styling

- A drop-down list added for easier navigation to recently visited folders

- **Yes/No** options changed to drop-down lists

- Bread crumb trails added to directly navigate to folder in a searched path

- Unneeded animation removed from some dialogs

- Use of the newer Explorer Style Tree View using folders and arrows

NOTE	If the DB Programming session is slow to respond, shut down any other programs—including antivirus programs—that may be running. If this action does not improve response speed, make sure the computer meets the minimum requirements. For more information about computer requirements, refer to the Specifications chapter in the *MiVoice Office 250 Installation Manual*.

Figure 1: Mitel DB Programming
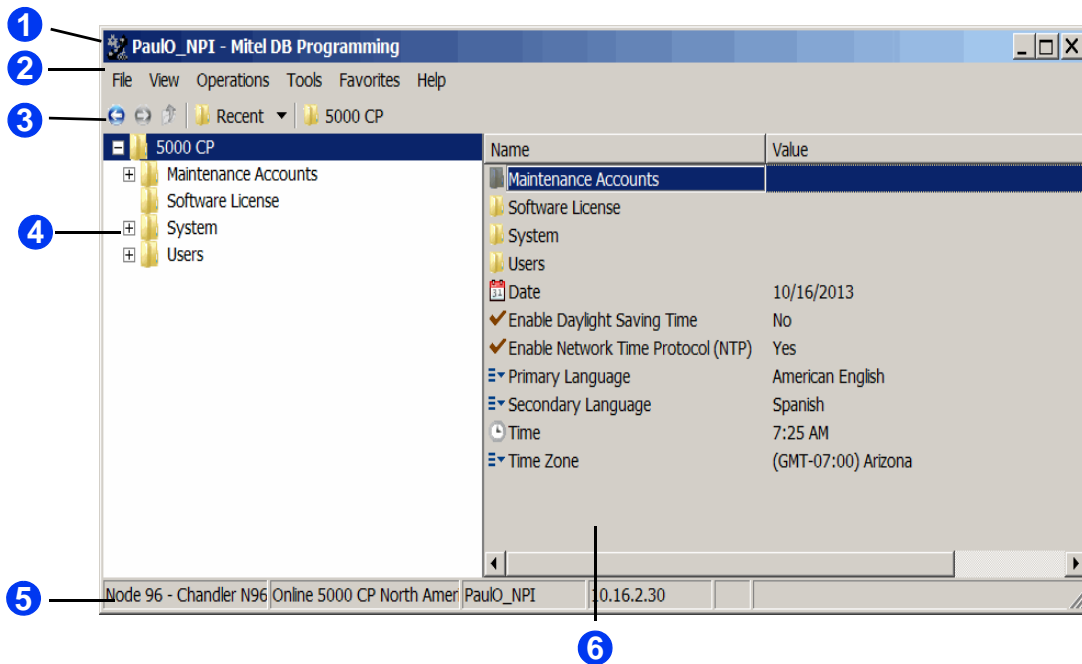
1	Header	Shows the title of the DB Programming application. See **page 26**.
2	Menu bar	Shows menu options. See **page 26**.
3	Toolbar buttons	Provide navigation options. See **page 28**.
4	Left pane	Shows Directory folders. See **page 28**.
5	Status bar	Shows system and connection information. See **page 31**.
6	Right pane	Displays options available for feature settings.

MITEL DB PROGRAMMING ELEMENTS

The following sections describe Mitel DB Programming elements.

HEADER

The header displays the title of the DB Programming application, "*<name>*[(Offline)] - Mitel DB Programming," where:

- the *<name>* indicates the name of the session.

- the (Offline) appears if the session is Offline (previously known as Local).

- Mitel DB Programming is the name of the application.

MENU BAR

Menus include the File, View, Operations, Tools, Favorites, and Help menus.

FILE MENU

This menu contains the Exit option only, which ends the DB Programming session.

VIEW MENU

The View menu includes the following options:

- **Toolbar**: Shows or hides the toolbar at the top of the window.

- **Status Bar**: Shows or hides the status bar at the bottom of the screen. The status bar shows the name of the session.

- **Back and Forward**: Allows you to move Back and Forward between views. You can also use the **Back** and **Forward** arrow buttons. See Figure 1 on **page 25**.

- **Collapse All**: Collapses all expanded folders in the left pane. You can also use the Collapse all button. See Figure 1 on **page 25**.

- **Classic Style**: Allows you to switch from Windows Vista folders style view to the Classic Style view showing icons in both panes.

- **Errors/Warning Window**: Enables or disables the errors and warnings window that appears at the bottom of the screen.

- **On-Line Monitor**: Enables or disables On-Line Monitor (OLM) mode. OLM is used for troubleshooting system problems. Use the OLM mode only when authorized by Mitel personnel.

- **System Device Info**: Displays all of the system devices. The Time Division Multiplexing (TDM) and IP devices appear at the top of the list, followed by the IP Networking resources, and then the system non-physical devices at the bottom.
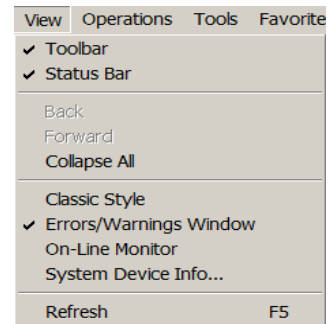
> | **NOTE** | 52xx/53xx IP phones awaiting PIN activation do not appear in the IP Device Status dialog box. See "Using PIN Activation for 52xx/53xx Phones" on **page 363** for details about the PIN Activation feature. |

- **Refresh**: Refreshes the screen.

OPERATIONS MENU

The Operations menu provides several options to help manage the system, as described in the following sections:

- Backup Operations on **page 1001**

- DHCP Server Options on **page 612**

- Error Information on **page 67**

- Export/Import Devices on **page 100**

- IP Device Status on **page 601**

- Software License Operations on **page 53**

- System Manager CA Certificate Upload on **page 107**

- Voice Processor Operations on **page 1001**

- Default Database on **page 66**

- Reset Call Processing Application on **page 73**

- Reset System on **page 73**

TOOLS MENU

The Tools menu includes the following options:

- **Configuration Wizard**: Used to configure new systems. See "Launching the Configuration Wizard" on **page 59** for instructions on how to launch the wizard. For complete information about the Configuration Wizard, refer to the *MiVoice Office 250 DB Programming Help*.

- **Networking Wizard**: Used to configure IP networking or T1/E1 PRI networking parameters See "Launching the Networking Wizard" on **page 99** for instructions on how to launch the wizard. For complete information about the Networking Wizard, refer to the *MiVoice Office 250 DB Programming Help*.

- **Resource Reservation Tool**: For detailed information about the Resource Reservation Tool, see **page 647**.

- **Shortcut Tool** (CTRL+T): Allows you to quickly jump to a specific programming area. For shortcut descriptions, refer to the *MiVoice Office 250 DB Programming Help* (press F1).

FAVORITES MENU

From the Favorites menu, you can add shortcuts to your most frequently visited folders. The menu also has options for sorting the list and deleting entries.

HELP MENU
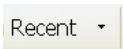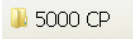
The Help menu includes the following options:

- **Help Topics**: Starts online Help.

- **About Mitel DB Programming**: Shows Mitel DB Programming version and copyright information.

- **About Mitel Database Programming**: Shows DB Programming version, DB Programming and CS5000SessMngr executable versions, and copyright information. For option descriptions and instructions, refer to Help (press F1).

- **About Mitel Software License**: Shows licensed features on the system. Additional software license information is available in the Software License directory. For option descriptions and instructions, refer to Help (press F1).

- **About Mitel Software Release and Packages**: Shows system software release and package information while connected in a remote-mode session. For option descriptions and instructions, refer to the Help (press F1). Not available for offline sessions.

TOOLBAR BUTTONS

Table 2 shows Mitel DB Programming toolbar buttons.

Table 2: Toolbar Buttons

ICON	NAME	DESCRIPTION
⬅ ➡	Back and Forward	Move backward and forward between views. You can also use the Back or Forward options in the View menu.
Recent ▾	Recent	This drop-down list allows you to jump to any of the twenty most recently visited folders. A maximum of 20 folders will be listed.
📁	Up	This button allows you to move up a level in the folder hierarchy. The button is disabled if you are at the top level of a folder.
5000 CP	Bread Crumb Trail	These buttons allow you to access any of the folders in the current folder path. Click on any folder name to go directly to that folder.

DIRECTORIES

The following subdirectories are shown in the Directory:

NOTE	Although the DB Programming window displays using the Windows Vista style folders view by default, you can revert back to the Classic Style. Windows Vista style controls (such as dialogs and buttons) remain. Some of the graphic representations in this guide are shown using the Windows Vista style.

- **Maintenance Accounts**: For more information about configuring system accounts, see "Maintenance Accounts" on **page 48**.

- **Software License**: For more information about software licensing, see "System Software Licenses" on **page 49**.

- **System**: The System directory features include most of the MiVoice Office 250 features and are discussed throughout this guide.

- **Users**: For more information about Users, see "Users" on **page 785**.

- **Voice Processor**: For more information about voice processor features, see "Voice Pro-cessor Features and Programming" on **page 835**.

SYSTEM DIRECTORY

Table 3 shows System subdirectories and folders. Option shortcuts appear in parentheses. For more information about shortcuts, see "Tools Menu" on **page 27**.

Table 3: System Subdirectories and Folders

SUBDIRECTORY	FOLDER
Controller (CNFG, CONF, SETU)	Circuit Modules
	Processor Module
	Processor Expansion Card
	Loop Start Ports
	Single Line Ports
	Digital Expansion Interfaces
	Error Thresholds (T1ERR)
Conference-Related Information	Meet-Me Conferencing
Devices and Feature Codes (DEV)	Assistants
	Extension Lists (EXTL or LIST)
	Feature Codes (FEAT)
	Hunt Groups (HUNT or UCD)
	IP Connections (IPC)
	Modems (MDM)
	Network Groups (NETG)
	Node IP Connection Groups (NIPCG)
	Nodes (NRG or NODE)
	Page Ports (PPORT)
	Page Zones (PAGE)
	Phantoms & Hot Desk Profiles
	Phones (STN)
	CO Trunk Groups (TRNK)
	Node Trunk Groups (NTG)
	SIP Peers
	Trunks (ILIN or INDT)
Echo Profiles	Voice Processor Echo Profile
	Conferencing Echo Profile
E-Mail Gateway (EMAIL)	--
File-Based MOH	--
Flags (MISC or SYS)	System-Wide Flags
Hunt Group-Related Information (HRI)	ACD Agent IDs (AGID)
IP-Related Information (IPRI)	Call Configurations (IPCC)

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Table 3:   System Subdirectories and Folders (continued)

SUBDIRECTORY	FOLDER (CONTINUED)
IP Settings (IPS)	Base Server/Processing Server Connection Settings
	DHCP Server Settings
	Web/SSH Settings
	Advanced IP Settings
	NTP Server Configuration
Maintenance (MAIN)	Call Cost (COST)
	Freeze Zones (FRZ)
	Major Reset Scheduling (MRS)
	Message Print (MPP or MPRNT)
	SMDR (SMDR)
Numbering Plan (NUM)	Area Flags (AREA)
	Classes Of Service (CLASS)
	Device Baseline Extensions (DEVBE)
	Dial Rules (DIAL)
	Emergency (EMER)
	Facility Groups (FAC)
	Home Area Codes (HOME) – USA only
	Route Groups (ARS)
	Toll Strings (TOLL)
	User Groups (USER) – USA only
Phone-Related Information (ERI)	Account Codes (ACCT)
	Application Profiles
	Applications
	Attendants (ATT)
	Audio Diagnostics (ADD)
	Flags (SFEA)
	Key Assignments - Keyset (KAS or KEY)
	Message Centers (CENT)
	Messages (MESG)
	Primary Attendants (PRI)
	System Forwarding Paths (SFWD)
	System Speed Dial (SSPD)
Reference Clock List (T1REF)	--
Sockets	--
System Manager (SYSM)	--
Timers and Limits (TIMR)	--
Trunk-Related Information (TRI)	Call Routing Tables (CALL)
	Loop Loss Test Settings
	Music-On-Hold Profiles
	SIP Gateways (SGTW)
System-Wide Fields (INTL)	--

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VOICE PROCESSOR DIRECTORY

Table 4 shows Voice Processor subdirectories and folders. Option shortcuts appear in parentheses. For more information about shortcuts, see "Tools Menu" on **page 27**.

Table 4:   Voice Processor Subdirectories and Folders

SUBDIRECTORY	FOLDER
Application-Related Information (AMISC or AMSC)	Audiotex Recordings (AUDIO or REC)
	Time Slot Groups (SLOT)
	Voice Mail Information (VMI)
	Auto Attendant Information (AAI)
Devices (ADEV)	Applications (AAPP or APPL)
	Digit Translation Nodes (DIGIT or TRANS)
	Extension IDs (AEX, EXID, or EXTID)
	Group Lists (AGL or GROUP)
	Mailboxes (AMAIL, AMB, or MAIL)
	Nodes (ANODE)
Fax-On-Demand (FAX or AFAX)	**Note:** This feature is not available when the system is using Unified Voice Messaging.
Mailbox-Related Information (MRI)	--
Maintenance (AMAIN)	--
Networking (ANET)	Timers and Limits
Report Parameters (AREP)	Automatic Report Generation
	Manual Report Generation
Timers and Limits (TIMR)	Timers and Limits
	DTMF Detection Information
	DTMF Generation Information
E-mail Synchronization	--
Voice Processor-Wide Parameters (ASYS or TALK)	--

STATUS BAR

The status bar at the bottom of the window shows:

- the description and number of the Node being programmed.

- the type of programming session.

- the region that the system is installed. The supported regions are North America and United Kingdom.

- the name assigned to the programming session.

It also displays status and error messages when applicable. The status bar also displays a description if you click and hold the cursor over the menu or menu option.

PROGRAMMING THE INACTIVITY TIMER

System Administration & Diagnostics and Database Programming now use the same timer for inactivity. The timer is programmed in the Options window on the Advanced Tab in System Administration & Diagnostics. For all versions or Database Programming, as soon as System Administration and Diagnostics version 6.0 is installed, all Database Programming versions (including all prior versions) will begin using the System Administration and Diagnostics timer for inactivity.

SELECTION WIZARDS

Any list in DB Programming that you use to add or move items through the shortcut menu uses a Selection Wizard (for example, adding members to a hunt group). The figure below shows an example Selection Wizard.
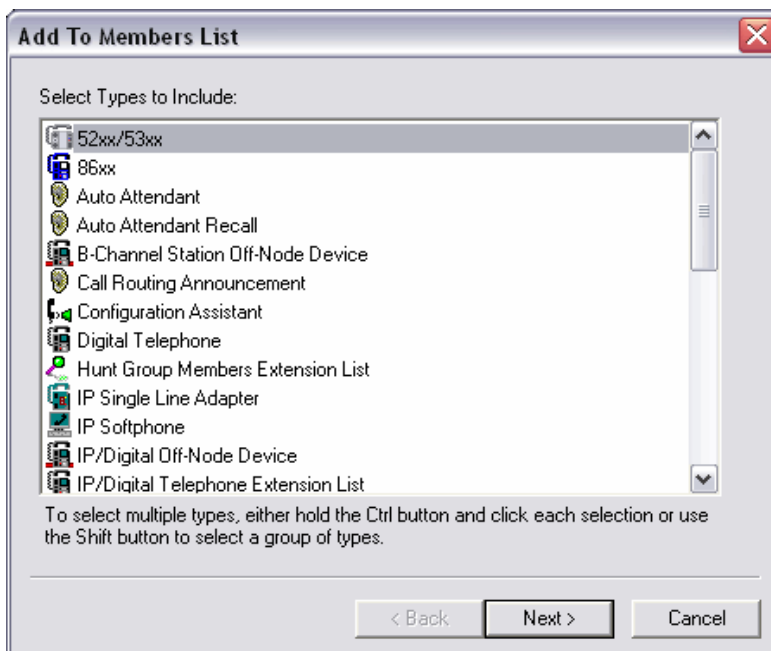
DB PROGRAMMING TIPS

The following sections describe general tips when using DB Programming.

VIEWING MITEL DB PROGRAMMING PANES

Remember the following when you view the left and right programming panes (see **page 24**):

- While programming, it is best to display the information in Details view to make the input fields visible.

- Click the column headings to sort a list of items. To sort files in reverse order, click the column heading once more. The tree view does not get sorted until you restart DB Programming.

- In many programming areas, click the right mouse button to see a menu of available commands.

- Several feature options instruct you to "click in a blank area of the right pane." If there is not a blank area available, you can also press CTRL+N.

CHANGING DISPLAYED INFORMATION IN THE PROGRAMMING WINDOW

For identification purposes, you can add information to the left pane of the system folders.

To add the information:

1. Right-click the option that you want to display in the folder name, as shown below.
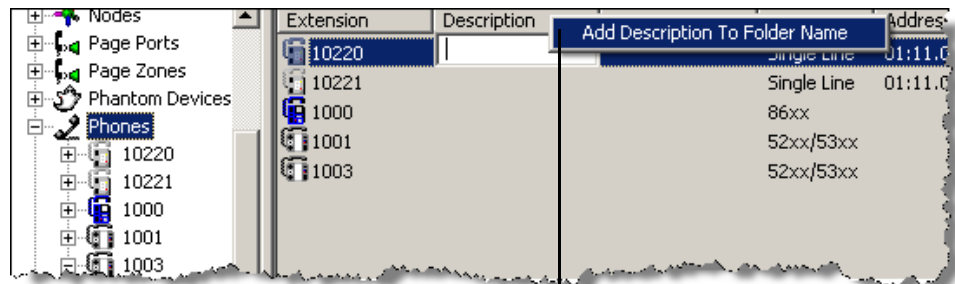
Right-click to add
a description to the left pane.

2. Click **Add <*option*> To Folder Name**. The option name appears in the folder name in the left pane, as shown below. To remove the field, repeat the procedure and select the option to remove the field. (The recommended limit is three fields.)

Description
Appears Here
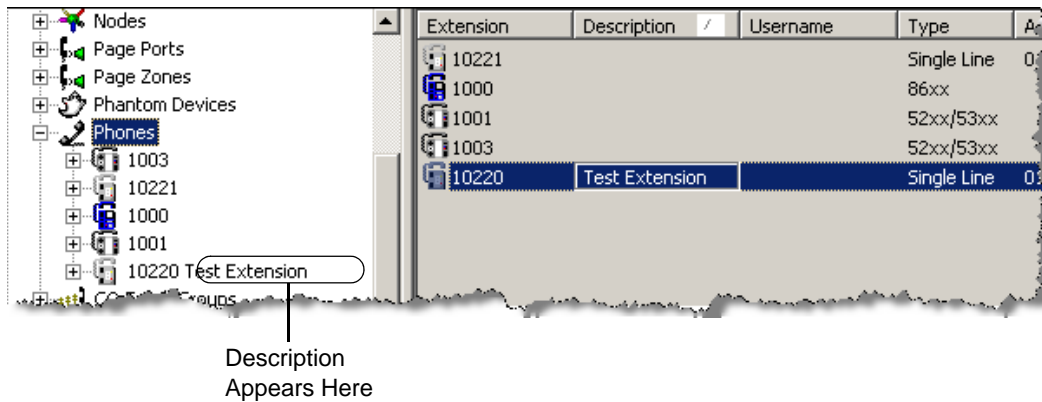
USING A KEYBOARD INSTEAD OF A MOUSE

You can use the keyboard to program feature options, without using a computer mouse.

Mouse Click Substitutions

The movement of the programming focus—that part of the screen where you want to make changes—is primarily controlled by the arrow keys and by the TAB and SHIFT + TAB keys. These keys move the programming focus right and left. The ENTER key takes the place of a single click, and CTRL + ENTER takes the place of a double-click.

- **Arrow Keys**: With the selection in the left pane, navigate using the arrow keys. Placing the selection on any item on the left pane automatically shows the contents of that item in the right pane, whenever applicable. For example, place the selection on "System" in the left pane and the computer displays the system contents in the right pane: Controller, Devices and Feature Codes, Flags, and so forth. At this point, the programming focus is on the words "System" in the left pane.

- **Expand (+) and Collapse (-)**: If the item in the left pane is expandable (which is indicated by a "+" sign to the left of the item), it is possible to expand it to show the contents in the left pane, by pressing the plus (**+**) key on the numeric keypad. The numeric keypad must be used for this function. Likewise, to collapse an expanded left pane list (indicated by a "-" sign to the left of the item), press the minus (**-**) button on the numeric keypad.

- **Tab/Shift to Change Focus Between Panes**: To shift the programming focus between panes, use **TAB** or **SHIFT + TAB** key. To select items, use **CTRL + ENTER**.

  To practice using these keys, do the following:

  a. Place the selection on System in the left pane. The System contents appear on the right.

  b. To select Devices and Feature Codes for programming, press **TAB**. Press the down arrow key. The selection in the right pane will move down to Devices and Feature Codes.

  c. To move from the right pane to the left pane, use **SHIFT + TAB**. You can use **TAB** and **SHIFT + TAB** to move to the drop down directory path window as well. This window gives a shortcut definition to your current programming selection and can be used to navigate to any point above your current location using the arrow keys.

d. At this point, your programming focus and selection is in the right pane on Devices and Feature Codes. If you want to select the **Devices** and **Feature Codes** area in the right pane, press **CTRL + ENTER**. (This is the equivalent of a double-click.) On the right side of the screen, Devices and Feature Codes expands showing Extension Lists, Feature Codes, Hunt Groups, etc. Notice that the programming selection has moved back to the left pane. This is because you have selected Devices and Feature Codes, but nothing below it. There is no selection in the right pane.

e. Now move the selection to the right pane using **TAB** and expand Extension Lists by pressing **CTRL + ENTER** while the selection is on it. Again, the selection will automatically move back to the left pane.

f. Put the selection back in the right pane, and use the arrow key to move down to Keyset and press **CTRL + ENTER** again. A list of Keyset Extension Lists appears in the right pane. Notice that the selection has again moved to the left pane. Move it back to the right pane using **TAB**.

- **CTRL + N to Add Items In The Right Pane**: To add an item to a list, press **CTRL + N**, to open the prompt that allows you to add to the list. Pressing the down arrow key selects this prompt. Pressing **ENTER** (equivalent of a single click) creates a new item.

- **CTRL + M to Edit An Existing Item**: To edit an existing item, use the arrow keys to select the entry, and then press **CTRL + M**. This opens the dialog box allowing you to explore (open) or delete the item.

- **ESC Closes Without Change**: Pressing **ESC** tells the system that you do not want to make a change. If a pop-up dialog box is present on the screen, it will be closed.

- **Changing A Flag**: Flags have two states (on/off, yes/no, and so forth). Note the following when changing flags:
  - When viewing a list of flags, press **TAB** to move the selection to the right pane.
  - Use the down arrow key to move to the desired flag. The selection is currently on the name of the flag, *not* the flag itself.
  - Move the selection to the flag to change it by pressing the right arrow key. You can see the focus of the selection move from the name of the flag to the state of the flag.
  - Press **ENTER** (single click) and the check box for the flag appears.
  - Change the status using the **SPACE BAR** to toggle between the choices.
  - When the check box shows the desired status, press **ENTER**.

- **Selecting From A Drop-Down List:**
  a. Move the programming focus to the list box, as described above for changing a flag.
  b. With the programming focus on the box, press **ENTER** to select this area to program. The box will "open up" and select the current Value.
  c. Use the up and down arrow keys to increment or decrement the value by 1. You can also type in a value directly.
  d. When the desired value is shown, press **ENTER**. An example of a drop-down list is the Controller programming folder.

Table 5 shows keystroke assignments.

Table 5:   Keystrokes for Navigating without a Mouse

KEY	ACTION
TAB SHIFT + TAB	Moves the selection between left and right pane or between lists (immediately below Menu Bar) and the left pane.
ALT + TAB	Switches between open programs in Windows.
Left or Right Arrow	Moves programming focus to left or right.
Up or Down Arrow	Moves programming focus up or down. Scrolls up or down in lists.
SPACE BAR	Changes status of a flag when the flag is selected for programming.
ENTER	Equivalent to a single click. Also used to accept a selected item in lists.
CTRL + N	Opens "Create" or "Add" dialog boxes.
CTRL + ENTER	Equivalent to a double-click.
ESC	When programming an item, it restores the original value (no change made) and removes focus from that item. If a list is open, it closes the list with no change.

FILTERING LISTS OF DEVICES

You can filter a list of devices when you add to a list (for example, adding to a list of members of hunt groups, adding to a list of phones with a particular class of service, adding to a list of audiotex recordings, and so forth). The figure below shows the "Add to Members List" dialog box.
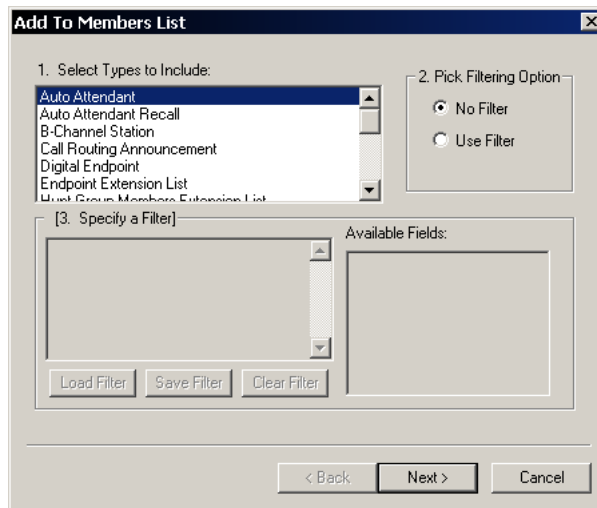
To filter a list:

1.   From the **Add to <list type>** dialog box, select the list type from the **1. Select Types to Include** list.

2.   Do one of the following:

Click **Next** to show a list of all devices of the types selected.

In the **2. Pick Filtering Option** area, select **Use Filter** to view filtering options. The following dialog box appears.
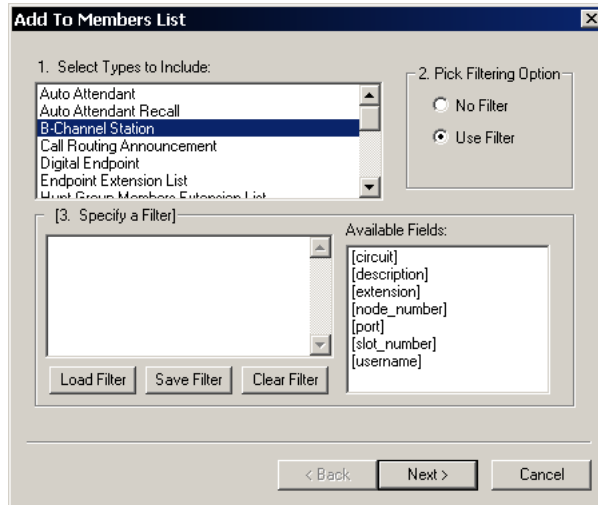
3. Select the filtering options from the 3. Specify a Filter and Available fields boxes. See "Using SQL Statements to Filter Lists" on **page 37** for information about using Structured Query Language (SQL) commands.

USING SQL STATEMENTS TO FILTER LISTS

To use the filtering option, a filter statement must appear in the "Specify A Filter" box. The statement must be presented in SQL format. **Table 6** shows SQL statement operators.

Table 6:   SQL Operators

<	Less than
<=	Less than or equal to (order of symbols must be correct)
=	Equal to
>=	Greater than or equal to (order of symbols must be correct)
>	Greater than
BETWEEN	A value that is greater than or equal to a specified lesser value and less than or equal to a specified greater value. For example 3, 4, 5 and 6 are BETWEEN 3 and 6.
NOT	Negates a statement.
IN	Same as OR
AND	Boolean operation indicating that two statements must both be true
OR	Boolean operation indicating that either of two statements must be true
?	Wildcard character for a single character
*	Wildcard character for any number of characters
LIKE	Used for equalities that are not exact matches. For examples, see the text.

The Available Fields box (see step 2 on **page 36**) lists the available SQL fields. Double-click a field to move it to the Specify a Filter box.

The "extension" field to be searched is in square brackets. The value to be matched in that field (the number 1234), is placed in quotes. The spaces on either side of the equal sign are not required but help make the statement easier to read.

You combine SQL commands and Available Fields to construct a SQL expression. For example, assume that you want to specify extension number 1234. The SQL expression to do this is: [extension] = "1234."

To place this expression into the Specify a Filter box:

1. Do one of the following:
   - Place the cursor in the box and type the expression.
   - Place the cursor into the **Specify a Filter** box, and then point at and double-click the desired field in the Available Fields box. The field appears at the cursor location in the filter specification box.

2. Type the remainder of the expression. If you press **Next**, the system presents a new box showing any phone with an extension of 1234. Remember that the device selected (phone) was specified in the Select Types To Include box.

You can use the equal (=) and less than (<) or greater than (>) commands to widen a search. For example, assume a database consists of eight phones with the extension numbers 1001 through 1008. To narrow the list, use the commands in **Table 7**.

Table 7:   Examples of SQL Statements

SQL STATEMENT	RESULTING LIST
[extension] < "1005"	1001, 1002, 1003, 1004
[extension] <= "1005"	1001, 1002, 1003, 1004, 1005
[extension] = "1005"	1005
[extension] > "1005"	1006, 1007, 1008
[extension] >= "1005"	1005, 1006, 1007, 1008

You can also use Boolean operators. **Table 8** assumes the same database of eight phones.

Table 8:   Examples of Boolean Operators for Searching the Database

SQL STATEMENT	RESULTING LIST
[extension] = "1005" OR [extension] = "1006"	1005, 1006
[extension] = "1005" AND [extension] = "1006"	Nothing found
[extension] NOT BETWEEN "1005" AND "1007"	1001, 1002, 1003, 1004, 1008
[extension] BETWEEN "1005" AND "1008"	1005, 1006, 1007, 1008
[extension] >= "1005" AND [extension] <= "1008"	1005, 1006, 1007, 1008

Notice how the last two expressions return the same result. You can use either the < and > symbols or you can use the Boolean expression BETWEEN for the same things.

The second expression returned nothing, even though it seems like it should have returned 1005 and 1006. The reason it returned nothing is because that expression requests a list of phones that have the extension number 1005 *and* the extension number 1006. To satisfy this expression, both numbers would have to be assigned to the same phone. This is not allowed by DB Programming. The correct expression would be to use the operator "OR," which would request a list of phones having the extension 1005 *or* the extension 1006.

Some of the fields do not require quotes around the values. These values are considered to be numbers, not string expressions. String expressions, such as extension numbers and user names must always be in quotes. Numeric expressions, such as node and port numbers, should not be placed in quotes. **Table 5-31** specifies the type of each field.

Table 9:   Types of Fields Requiring Quotation Marks for a Search

FIELD	TYPE	REQUIRES QUOTES?
[circuit]	Number	No
[description]	String	Yes
[extension]	String	Yes
[node_number]	Number	No
[port]	Number	No
[slot_number]	Number	No
[username]	String	Yes

If you forget which type it is, an error message alerts you that you have a "Data type mismatch," and you must return to the statement and enter it correctly.

USING SQL WILDCARD CHARACTERS

Wildcard characters that can be used for filtering are:

- Asterisk (*), which means any character or characters are accepted.

- Question mark (?), which is used to indicate a match of a single character.

Because filtering uses statements consisting of letters as well as numbers, the equal sign means that an exact match must be present. Therefore, the expression *<username> = "FRED*"* only returns a user name that has the letters F, R, E, D, and an asterisk; that is, an actual asterisk as the fifth character of the username. It would not return "FREDDY" or "FRED SMITH." To use the wildcard, the SQL statement must tell the search engine to locate something that does not exactly equal the search string specified in the quotes. To do this, use the SQL command "LIKE."

The expression *<username> LIKE "FRED*"* returns user names of "FRED," "FREDDY," "FRED SMITH" or any other username that starts with the letters F, R, E, D.

Similarly, the question mark is used to indicate only a single character space is a wildcard. **Table 10** shows some additional examples.

Table 10: SQL Statements and Search Results

SQL STATEMENT	SEARCH RESULT
[username] = "FRED?"	FRED?
[username] LIKE "FRED?"	FRED plus any single character
[username] LIKE "FRED*"	FRED followed by any or no additional characters
[username] LIKE "FRED??"	FRED followed by any two characters
[username] LIKE "FRED?*"	FRED plus at least one more character
[username] > "b"	Any username except those beginning with the letter A
[username] < "a"	Any username that would appear before those beginning with the letter A, (for example, blanks)
[extension] LIKE "12??"	Extensions in the range 1200 through 1299 inclusive

COMBINING SQL COMMANDS

You can combine search patterns using the Boolean operators previously discussed, as shown in **Table 11**.

Table 11: Combined SQL Statements and Search Results

SQL STATEMENT	SEARCH RESULT
[username] = "FRED" AND [extension] = "1234"	Only 1234 with a username of FRED
[username] = "FRED" OR [username] = "WILMA"	Any phone with a username FRED or WILMA
[username] = "FRED" OR [extension] = "1234"	Extension 1234 as well as any with FRED as the username
[extension] LIKE "12*" AND [username] LIKE "F*"	Extensions beginning with the digits 1 and 2 that have usernames beginning with the letter F
[extension] < "1234" AND [username] = "FRED"	Extensions with a number lower than 1234 that also have a username of FRED
[bay_number] = 1 AND [extension] >= "1000"	Phones on module number 1 having extension numbers 1000 and higher

USING THE CLEAR FILTER, SAVE FILTER, AND LOAD FILTER COMMANDS

Three command buttons appear below the Specify A Filter box:

- **Clear Filter**: Removes the entire entry in the Specify A Filter box.

- **Save Filter**: Saves the filter for future use. There may be instances where you have a complex, or a commonly-used search that you want to use over again. When you have a search statement that you want to keep for future use, save it for later retrieval. To do this, select **Save Filter** when the search statement is in the Specify A Filter box. A dialog box appears prompting you to specify a location and file name. The file is saved in that location with an "FLT" extension.

- **Load Filter**: Finds a previously saved search string and restore it to the Specify A Filter box.

DATABASE UTILITIES

MiVoice Office 250 utilities are incorporated into the System Administration & Diagnostics application. For information about how to launch the following utilities, refer to the *System Administration & Diagnostics Guide*, part number 550.8125:

- Database Converter Utility

- MOH Converter Utility

- DB Test and Repair Utility

- Upload Utility

Note that the Diagnostics Monitor utility is no longer available or needed in v4.0 or later DB Programming, as its functionality is incorporated into the System Administration & Diagnostics application. For complete information about utilities, refer to the appropriate utility Help.

Chapter 3

SYSTEM MANAGEMENT

INTRODUCTION

This chapter describes features that you can use to manage and maintain your MiVoice Office 250. System management options include:

- Security Enhancements on **page 45**

- Maintenance Accounts on **page 48**

- System Software Licenses on **page 49**

- Backup Operations on **page 60**

- Launching the Configuration Wizard on **page 59**

- Backing Up the Database on **page 64**

- System Error Information on **page 67**

- System Maintenance Options on **page 68**

- System Resets on **page 73**

- on **page 89**

> ## NOTICE
>
> **System Performance**. Perform the following system management options after hours, when system usage is at a minimum.
> - Imports (with a large database, especially with off-node devices). For example, performance can also be affected when other nodes export to the local node.
> - DB backup saves (with a large database, especially with off-node devices).
> - Paging to a large number of phones.
> - All-ring hunt groups to a large number of phones.
> - Abusive/intention actions (user runs a "die macro" on their phone).
> - Change extension logic (DB Programming performs a batch modify/delete extension (the larger the database, the slower the action).

SECURITY ENHANCEMENTS

The v4.0 and later software release includes a number of security enhancements related to system accounts, passwords, password policies, encrypted connections, security key strength, certificate management, and Secure Socket Layer (SSL) usage. See the following sections for details.

SYSTEM ACCOUNTS

Beginning with 5000 Release 6.0, System Accounts has been renamed Maintenance Accounts. This folder shows only the Administrator and Support accounts for Database Programming. No other accounts are listed in this folder. User accounts that have been given administrative-related access privileges will have an Administrator-Related Information folder in their respective Users folder.

The support account is disabled by default, and it can only be enabled by an admin account user (for example, when a system administrator requests technical assistance from Mitel Technical Support). This feature gives the system's designated administrator(s) control over who can log on to the system. See for details.

PASSWORDS AND PASSWORD POLICIES

The default password for both the admin and support accounts described above is "itpassw", but it must be changed during the first logon attempt, as described later in this section.

The DB Programming "System Accounts" folder (previously called the "Passwords" folder) that allows different levels of DB Programming access now requires a user name and password pair instead of just a password.

There are six password policies applicable to the system:

- **Policy 1:** All default passwords for "admin" MUST be changed on first log on.

- **Policy 2:** The default password for "support" MUST be changed when the admin enables it for the first time.

- **Policy 3:** New passwords MUST be different from the default password.

- **Policy 4:** The minimum password length allowed is 6.

- **Policy 5:** The password MUST contain at least one uppercase character, one number, and one special character.

- **Policy 6:** The minimum password length MUST be X, where X is configurable with a range of 6 to 24.

The first four policies are mandatory and will be applied by the system on every connection. Policies five and six are configurable by the administrator.

Changing Passwords

When the System Administration & Diagnostics application connects to a MiVoice Office 250 for the first time using the default "admin/itpassw" credentials, the system prompts the administrator to change the password for the admin account. (The new password is stored in

System Administration & Diagnostics so that subsequent online sessions will not prompt for a password.) Likewise, when the support account is enabled for the first time, the administrator is prompted to change the password for this account.

After restoring a database to a system, the username and password stored in System Administration & Diagnostics may need to be changed to match username and password in the restored database in order to successfully connect to the system.

NOTE	If the system is configured with a Processing Server (PS-1), all password changes must be synced up between the Base Server and the PS-1. Passwords cannot be changed if the Base Server and the PS-1 are not connected.

ENCRYPTED CONNECTIONS

With v4.0 or later software, the following applications use an encrypted connection for inbound communication with the MiVoice Office 250:

- System Administration & Diagnostics Tools/Utilities: server TCP port 44000 and 443 (configurable)

- Database Programming: server TCP port 44000 (configurable)

- Web Browsers: server TCP port 443 (configurable)

These applications use either SSLv3 or TLSv1 for the encrypted channel.

Users of these applications will now receive certificate notifications as the client application attempts to verify the identity of the connected device.

INCREASED SECURITY KEY STRENGTH

All public and private keys generated by or for the system use the PKCS #1: RSA Cryptography Standard (an implementation of Public-Key Cryptography Standards based on the RSA Laboratories algorithm) and are at least 1024 bits in length.

CERTIFICATE MANAGEMENT

The system uses the Internet X.509 Public Key Infrastructure (PKI) Certificate Management Protocols for all relevant aspects of certificate creation and management.

SSL PORT USAGE

Instead of port 4000 (a non-SSL port), these applications use port 44000 (an SSL port):

- System Administration & Diagnostics

- DB Programming

- Message Print

- System Monitor

- SMDR application built into System Administration & Diagnostics

System and desktop OAI applications and System Manager continue to use port 4000. An attempt to connect via SSL on a non-SSL port (4000 by default) will result in connection failure. Likewise, an attempt to connect without SSL on an SSL port (44000 by default) will result in connection failure.

MAINTENANCE ACCOUNTS

Beginning with 5000 Release 6.0, System Accounts has been renamed Maintenance Accounts. This folder shows only the Administrator and Support accounts for Database Programming. No other accounts are listed in this folder. User accounts that have been given administrative-related access privileges will have an Administrator-Related Information folder in their respective Users folder.

A User's access can be changed by opening the Users folder and right-clicking on the user and selecting Edit User. The User Creation Wizard dispalys to allow any changes for the user.

The following account types are available in DB Programming:

- **Administrator** (username = admin): Intended to be used by system administrators and field technicians. This account has all of the functionality to program and maintain DB Programming. The admin account user can edit the admin account password in the Maintenance Accounts folder, and also can add or remove other accounts, enable/disable the support account, edit descriptions and access rights, and edit other account passwords. *This is the only system account enabled in a default database. The default password is set to itpassw.*

- **Support** (username = support): Intended to be used by Mitel technical support personnel. The support account is disabled by default, but when it is enabled, it has nearly all of the functionality of the admin account. The only limitation the support account has is that the support account user can edit only the support account password in the Maintenance Accounts folder. The support account user cannot change access rights or make changes to the admin or other accounts. The purpose of the support account is to give temporary system access to technical support personnel without having to share the admin password. *The admin account user can enable the support account with a temporary password and disable the support account when finished. The default password is set to itpassw. This account must be enabled prior to getting any assistance from Mitel Technical Support.*

The password can be edited for both accounts, and support accounts can be enabled/disabled.
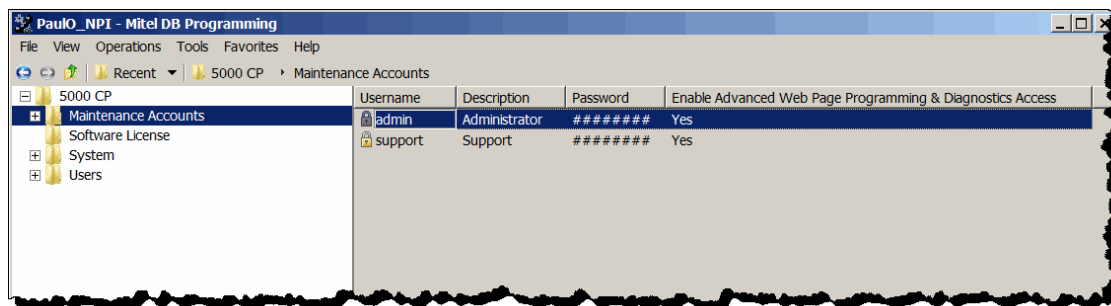
SYSTEM SOFTWARE LICENSES

Once connected to DB Programming for the first time, you must upload the software license. If you do not upload the software license, the system considers the license as "invalid" and issues Alarm 125.

For a complete list of alarm messages, refer to the *Message Print Diagnostics Manual*, part no. 550.8018.

IMPORTANT	With regard to the security device and SSL: • The serial number is stored on a serialized compact flash-type card or a USB security key. The license must match the serial number on this security device or the upload will fail. If for any reason you need to replace the security device, make sure you upload the latest license file *before* replacing the device. Attempting to upload a license file *after* changing the security device results in an error and the system will not function. • If the security device is not installed when the system reboots, the software license will need to be uploaded.

This section contains the following information:

- **Obtaining a Software License** below

- **Viewing Software Licenses** below

- About Mitel Software License on **page 53**

- Uploading Software License on **page 53**

- Comparing or Uploading a Software License on **page 54**

- Changing Records on **page 55**

- Changing Licenses on **page 55**

- Downgrading Licenses on **page 56**

- Troubleshooting Licensing Issues on **page 56**

OBTAINING A SOFTWARE LICENSE

In general, the site should already have a software license available. If necessary, however, you can "build" a license to meet the customer's needs. All licensing is handled through the Application Management Center (AMC). For information on the AMC, refer to the Help or other documentation available at

For information obtaining software licenses, refer to the AMC Help or the Mitel Web site (http:/ /edocs.mitel.com).

VIEWING SOFTWARE LICENSES

You can view the software licenses and values currently uploaded to the system, including features common to all licenses. Software licenses are read-only.

To view software licenses:

Select **Software License**. Software licenses are shown in the right pane.

The detailed information about software license features is shown in **Table 12**. The values shown are for offline mode. In online mode, the values match the software license that is loaded. If there is no license loaded or the current license on the system is invalid, the fields display with a red "X."

You can create as many IP phones as the system can support (up to 250). Call Processing uses the various Phone licenses to determine which phones come online. Note that licensing is per online phone, not per phone usage.

Beginning with the v5.0 software release, Primary Rate Interface (PRI) capability no longer requires a premium feature license. All enabled ports on the Single and Dual-Port T1/E1/PRI Modules (T1M and T1M-2) may be configured for the PRI protocol without additional charge. (Note that a license is still required to enable the second port on a T1M-2).

NOTICE
You must upgrade system software *before* loading the license.

Table 12: Software License Descriptions

FIELD	DESCRIPTION
System Type	Displays the system type, MiVoice Office 250.
ACD Hunt Group	Indicates whether or not the ACD Hunt Groups software license is uploaded to the system.
Additional T1/E1/PRI Ports	Indicates how many additional Dual T1/E1/PRI port licenses are uploaded to the system.
Agent Help	Indicates whether or not the Agent Help software license is uploaded to the system.
Analog Voice Mail Hunt Group	Indicates whether or not the Analog Voice Mail Hunt Group software license is uploaded to the system.
Category A Phones	Indicates how many Category A phone licenses are uploaded to the system. Category A license includes the following phones: • Mitel 8600, 8622, 8662, and 8690 phones that are in ITP (Inter-Tel Protocol) mode • OAISYS Net Phone (third-party SIP phone)
Category B Phones	Indicates how many Category B phone (Mitel 8602 IP softphone) licenses are uploaded to the system.

Page 1 of 3

Table 12:   Software License Descriptions (continued)

FIELD	DESCRIPTION
Category C Phones	Indicates how many Category C phone licenses are uploaded to the system. Category C license includes the following phones: • Mitel 8600, 8622, 8662, and 8690 phones that are in SIP mode • Mitel 8660 phone • IP Single Line Adapter (SLA) • Mitel 8664, 8665, and 8668 phones • Mitel 8601 SoftPhone for Pocket PC • Cisco® 7960 SIP Phone (third-party SIP phone)
Category D Phones	Indicates how many Category D phone (52xx\53xx phone) licenses are uploaded to the system.
Category E Phones[1]	Indicates how many Category E phone (digital telephone) licenses are uploaded to the system.
Category F Phones	Indicates how many Category F phone licenses are uploaded to the system. Category F license includes the following SIP phones: • Mitel Unified Communicator (UC) Express SIP Softphone • Mitel 5610 Cordless Handset and IP DECT (Digital Enhanced Cordless Telecommunications) Stand • Mitel 5603/5604/5607 • Mitel UCA SIP softphone
Desktop Interface	Indicates whether or not the Desktop Interface software license is uploaded to the system.
Dynamic Extension Express	Indicates whether the Dynamic Extension Express software license is uploaded to the system. For more information about Dynamic Extension Express, see "Dynamic Extension Express" on **page 807**.
File-Based MOH Sources	Determines how many File-Based MOH sources can be used on the system.
Hot Desking	Indicates whether the Hot Desking software license is uploaded to the system.
IP Gateway Trunks	Indicates how many IP gateway trunk licenses are uploaded to the system. If you are upgrading your system, your current IP Gateway Trunk licenses will convert to the SIP Trunks license.
IP Networking	Indicates how many IP networking trunk licenses are uploaded to the system.
Meet-Me Conferencing	Indicates whether the Meet-Me Conferencing software license is uploaded to the system.
Remote Automatic Call Distribution Hunt Groups	Indicates whether or not the Remote Automatic Call Distribution Hunt Groups software license is uploaded to the system.
SIP Trunks	Indicates how many SIP trunk licenses are uploaded to the system. If you are upgrading your system, your current IP Gateway Trunk licenses will convert to the SIP Trunks license.
SIP Voice Mail Ports	Determines how many SIP peer voice mail ports can be used on the system.
System OAI Events	Indicates whether or not the System OAI Events software license is uploaded to the system.
System OAI Third Party Call Control	Indicates whether or not the System OAI Third Party Call Control software license is uploaded to the system.
Virtualized PS-1 Support	Indicates if a Virtualized PS-1 license is installed on the system.

Page 2 of 3

Table 12:   Software License Descriptions (continued)

FIELD	DESCRIPTION
Voice Processor Messaging Networking	Indicates whether or not the Voice Processor Messaging Networking software license is uploaded to the system.
Unified Voice Messaging Ports	Indicates how many Unified Voice Messaging port licensed are uploaded to the system.
	Prior to v5.0, the embedded Unified Voice Messaging application (previously called Basic Voice Mail) was limited to a maximum of 16 voice mail ports. With v5.0 or later, UVM can be licensed for up to 24 or 32 voice mail ports. The original CS Controller Base Server supports up to 24 ports. The HX Base Server and PS-1 equipped platforms support up to 32 ports.
Unified Voice Messaging Blackberry® Integration	Indicates whether the UVM Blackberry Integration software license is uploaded to the system.
Unified Voice Messaging E-mail Synchronization	Indicates whether the UVM E-Mail Synchronization software license is uploaded to the system.
User Web Portal	Indicates whether the User Web Portal license is uploaded to the system.

Page 3 of 3

1. Digital Expansion Interface #1-4 licenses, if present, correlate to 48 digital telephone licenses upon conversion.

The MiVoice Office 250 requires a version-specific license. You can install and run DB Programming, but you are prompted to upload the license file. For information about uploading a software license and for a list of features that require licenses, see "Uploading Software License" on **page 53**.

ABOUT MITEL SOFTWARE LICENSE

The About Mitel Software License dialog box shows current licenses allocated for your system, as shown in **Figure 5**. For more information about this dialog box, refer to the *MiVoice Office 250 DB Programming Help*.

Figure 5: About Mitel Software Licenses
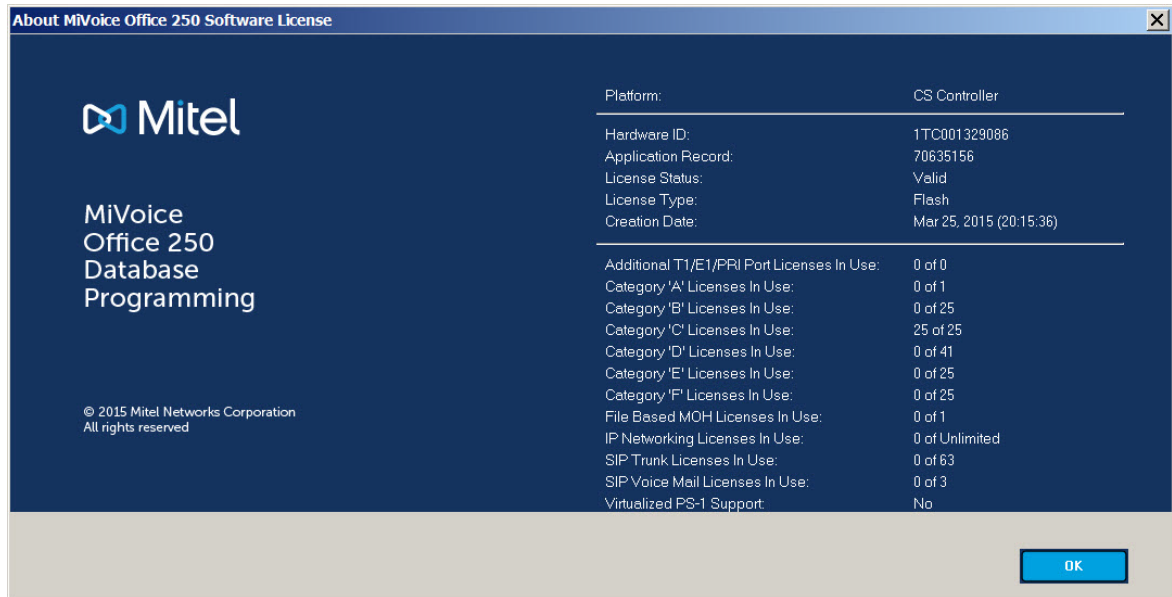
UPLOADING SOFTWARE LICENSE

With the v3.2 or later software release, the system supports licensing through a serialized compact flash-type card rather than through a USB key. (This applies only to systems not equipped with a PS-1. Systems with a PS-1 must still use a USB key for licensing.)

All features currently licensable using a USB key can be made available for licensing using a compact flash-type card. The license file from the AMC can be associated with either medium– but not both concurrently. For details, refer to the Product Description chapter in the *MiVoice Office 250 Installation and Adminstration Manual*.

To upload a software license:

1. From the DB Programming menu bar, select Operations – Software License Operations – **Upload Software License**.

2. Enter the path and name of the license file (`.isl`) or click **Browse** to search for the file.

3. Click **Start**. The system compares the new software license to the existing license, if applicable.

   NOTE   If you click Start before selecting a license (`.isl`), a message displays, informing you that you have not identified a file name.

4. Click **Finish** at the Compare Software License screen. If you upload a software license that does not support your current programming, a warning message appears.

COMPARING OR UPLOADING A SOFTWARE LICENSE

When you upload a software license, the system checks the Unified Voice Messaging (UVM) ports available in the new license and the current resources allocated to UVM. If the current resources allocated exceed the number of licensed UVM ports, you are prompted to change the resource allocation.
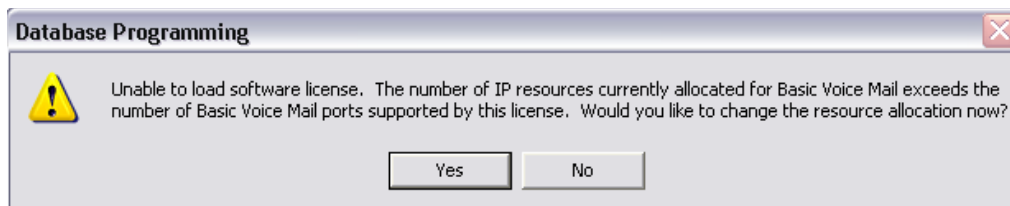
If you click **Yes**, the view is changed to the System – Controller – IP Settings – IP Resource Allocation folder so you can change the IP resources allocated to UVM.

If you click **No**, you return to the folder you were viewing, prior to the software license upload, and the upload is canceled.

You must license additional UVM ports. If you downgrade a license to support fewer UVM ports, DB Programming performs a check on the Time Slot Groups (see "Time Slot Groups" on **page 868**). If the Time Slot Groups currently have a Maximum Channel Allocation greater than the new license supports, a warning message appears, preventing you from uploading the license. You are prompted to go to the Time Slot Groups folder and change the Maximum Channel Allocation fields to a value that is supported in the new license. If you do not go to the Time Slot Groups or you click **Cancel**, you are returned to DB Programming and the new license is not loaded.

UVM ports, IP Phone Categories, and Node Capacity are displayed in the Compare License dialog box.

When a new license is loaded, the dialog box showing the comparison between the old license and the new license shows the License Type. The available license types are:

*   USB (for the old software keys)

*   Compact Flash (for the new way of storing a license)

*   Unknown (for old licenses that did not have this field)

The following illustration shows an example of the Compare Software License dialog box that appears during the software license upload process.
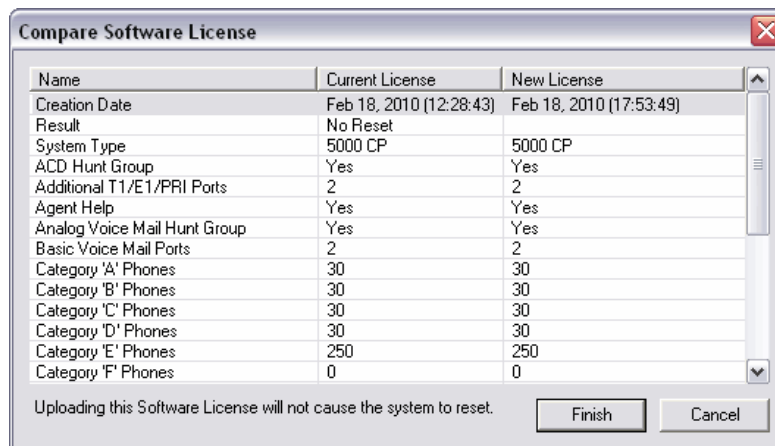
SYSTEM MANAGER CA CERTIFICATE UPLOAD

If a node will connect to System Manager, you must upload the Certification Authority (CA) certificate. The certificate allows the node to use Secure Socket Layer (SSL) for a secure connection.

> **NOTE** You cannot upload the CA certificate while in local mode.

To upload the CA certificate:

1. From the DB Programming menu bar, select Operations – **System Manager CA Certificate Upload**.

2. When prompted, click **Browse** and locate the CA certificate (usually at `C:\Program Files\Mitel\System Manager\CA`). Certificates have a `.cer` extension.

3. Click **Start**. The CA certificate is uploaded.

For details about using System Manager, refer to the *System Manager Installation Manual*.

CHANGING RECORDS

If problems occur with the security device, you may have to change a record from one system to another. See Table 14, "Invalid License Upload Errors," on **page 57** for a list of error messages.

For information on changing records, refer to the Application Management Center (http://www.ebiz.mitel.com/amclogin.jsp).

CHANGING LICENSES

If necessary, you can upload a new license that provides a different set of features. If the new license, however, supports new capacities, you must make sure that you have the appropriate hardware installed.

To change a license, obtain a new license, as described on **page 49**, then upload the license. The system compares the new license to the existing license.

In this screen, the areas of the license that are changing are selected. The status line indicates whether the system requires a reset once the license is uploaded. After you have verified you want to upload the new license, click **Finish**. The new software license is loaded, and the old software license is stored on the system. The system may require a reset.

DOWNGRADING LICENSES

Mitel strongly recommends against downgrading a software license.

Uploading a software license where the number of licenses is different than the current license count. Before accepting the new software license, DB Programming displays where the license field in question is different than the existing license field and warns that the system will reset.

When uploading a new software license, upload the latest generated `.isl` license file to ensure that the software license does not downgrade.

If a system has continuous license key failures, Alarm 132 is posted on all administrator phones. After the system successfully recognizes the license key, upon the next reset, the system checks for a valid software license.

If the license key is functional but the system does not have a software license, the system posts alarm 125 every 5 minutes, then resets after 4 hours.

TROUBLESHOOTING LICENSING ISSUES

Table 13 summarizes troubleshooting strategies recommended for resolving licensing discrepancies.

Table 13:   Licensing Troubleshooting Tips

Symptom	Possible Cause	Possible Solution
The system resets due to a "Major Reset Due To an IP Phone Licensing Error."	The user has uploaded a software license that causes the number of online Advanced IP phones to exceed the number of licensed Advanced IP phones.	Mitel recommends *not* downgrading a software license. Before committing the new license, a message pop-up in DB Programming indicates that the system will reset if the new license is installed.
Alarm 127 appears on the administrator phone display and on the Base Server LCD display.	An IP phone cannot come online because the appropriate license is not available.	The customer needs to upload a license to support the phone model type.

Invalid License Upload

A software license is created for a specific system type, so if you attempt to load a software license that does not match the system type, an error message displays. If you attempt to load an invalid license, the following error message appears. This event may occur when using the Upload Software License feature, which is accessed by selecting Operations and then Software License Operations in DB Programming.
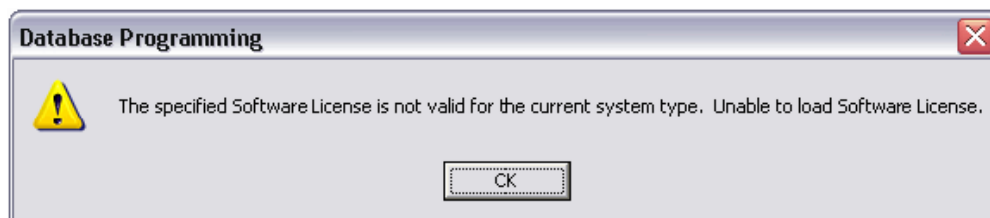
If the license is invalid, an error message is displayed, indicating why the upload failed. A license may be considered invalid for a variety of reasons, as described in **Table 5-2**.

Table 14:   Invalid License Upload Errors

Error Message	Possible Cause	Corrective Action
Corrupted Software License. Unable to load Software License.	The software license is corrupted.	Connect to the Mitel Applications Management Center (AMC) and download the file again. If this error occurs again, contact the Software Maintenance Center.
		Note that when using the virtual PS-1, a "Security key is either invalid or missing" error may appear if the USB device is not mapped to the correct virtual server. See the *MiVoice Office 250 Installation and Adminstration Manual* for details.
The specified Software License is not valid for this specific CPU. Unable to load Software License.	The serial number on the does not match the number in the license.	Load a license that has the correct serial number.
The specified Software License has expired. Unable to load Software License.	The software license is no longer valid or has expired.	Load (or request) a new Software License.
Too many devices equipped. Unable to load software license.	The current database is programmed for more devices than the software license allows.	Either use local mode to unequip some devices or upgrade the capacity limit on the license.
Software License version is not supported. Unable to load Software License.	The license you have attempted to load is for a different version of system software.	Either install the correct version of system software or upload a software license that specifies the correct system version.
Software License is already loaded. Unable to load Software License.	A valid software license is already loaded on the system.	If the software license you are attempting to load is the correct license, start a new DB Programming session with a default database. Then upload the software license before restoring the database.
Software License is not valid for the current CPU type.	The CPU type identified in the software license does not match the CPU type detected in the system.	Upload a license that identifies the installed CPU type.

Table 14:   Invalid License Upload Errors (continued)

Error Message	Possible Cause	Corrective Action
Security key is either invalid or missing.	The software license serial number does not match the serial number on the security key, or the security key is not present on the server.	Make sure the security key is present and properly seated in the USB port. If the key is present, make sure the license reflects the correct serial number. If necessary, transfer the license to the correct serial number (see **page 55**).
The specified Software License is not valid for this software version. Unable to load Software License.	The software license product version does not match the version installed on the system.	Either install the correct system version or upgrade the license to reflect the correct system version.
The specified Software License encountered an unknown upload error. Unable to load Software License.	An unidentified error occurred.	Contact the Software Maintenance Center.

Page 2 of 2

LAUNCHING THE CONFIGURATION WIZARD

The Configuration Wizard allows you to quickly configure a system with enough basic information to get the system up and running. Verify the system type to make sure it matches the system you have. If there is a difference, either the hardware is incorrect or the license type is incorrect.

The Configuration Wizard opens when Database Programming is started for a default system.

A **Show this wizard at DB Programming startup** check box on the main welcome page allows you to have the Configuration Wizard start whenever Database Programming starts up.

You can program the following settings using the Configuration Wizard:

- IP Settings

- Email Gateway Configuration

- Date and Time Settings

- Controller Expansion Module Installation for the following modules:

  - Digital Expansion Interface Module

  - Voice Processor

- Users/IP Phones (a Create User wizard appears when creating a user)

For complete information about the Configuration Wizard, refer to the MiVoice Office 250 DB Programming Help.

To start the Configuration Wizard:

In addition to having the Configuration Wizard starting when opening Database Programming, from the DB Programming menu bar, select **Tools**, and then select **Configuration Wizard**. The Configuration Wizard Welcome screen appears.

BACKUP OPERATIONS

Backup operations now include the ability to to perform a bundled save where all available system data (Voice, HTML Applications, and File-based Music on Hold files) will be retreived from the system and saved in a backup file at a specified location.

Beginning in Release 6.0, the Database Operations menu was renamed Backup Operations. Backup save and restore options are available under the **Operations→Backup Operations** menu.

After the System and Voice Processor databases have been programmed for the first time, perform a database save. Store this database in a safe place so it can be used should the current the database become corrupted and must be re-entered. This saved database can be used as a starting point for the re-entry. In addition, keep a current copy of the database that you can update every time you make changes.

SAVE BACKUP

Four options are provided when saving a backup.

IMPORTANT	Performing a Save or Restore operation will cause the system to slow down. If at all possible, these operations should not be performed during normal business hours when the system is being used to place and receive calls.

To save a database backup:

1. Go to **Operations→Backup Operations→Save Backup. . .**.
   The *Save Backup* window displays.

2. In the **I want to save** drop-down list, select from one of the following options:

   - **Database to a PC**: This is the selected default option. Only the system database, not any system data, will be saved to the backup file as in pre-6.0 database saves.

   - **Database to a PC and voice data elsewhere**: The database is saved in the backup folder in a backup file. The Voice Data is saved at an alternate location (USB thumb drive on the system or network drive for EVM/PS-UVM). See "Saving and Restoring Voice Processing Databases" on **page 1001** for more details.

   - **Database to the system**: This option will save the CP database on the system for recovery following a CP reset.

   - **Everything to a PC**: This option will perform a bundled save operation where all available system data (Voice, HTML Apps, FB-MOH Files) will be retrieved from the system and saved in the backup file at the specified location.

NOTE	Database Programming will use the Web Listening Port defined in the System Connection to save the additional data. If the Web Listening Port is not open, a warning message dispalys.

3. In the **Save to** filed, enter the location to where the backup will be saved, or click the ellipses at the end of the field and navigate to the location.

4. Select the **Encrypt data** check-box to encrypt the backup.

5. Click **Start**.

The "Backup Summary" will list what will be restored once selections are made.

RESTORE BACKUP

This option is located under **Operations→Backup Operations**. As part of the bundled save/restore feature, Restore Backup will perform a bundled restore operation of all available system data (Voice, HTML, Applications, and File-Based Music on Hold files) from the backup file at a specified location.

NOTE	Consider the following technical consequences when using a modem connection: **1. Possible Database Corruption**. Poor line quality may cause data transmission problems when the modem connection exceeds 19200 baud. For this reason, Mitel recommends *not* using the modem to restore the database. If you attempt a restore using the modem, the database may become corrupt. **2. Possible Modem Reconfiguration**. If the Database Restore feature is used during a remote programming session, all calls are disconnected except the modem connection. Before restoring the database, make sure the modem connection will *not* be reconfigured during the restoration.

IMPORTANT	Performing a Save or Restore operation will cause the system to slow down. If at all possible, these operations should not be performed during normal business hours when the system is being used to place and receive calls.

To restore a database backup:

1. Go to **Operations→Backup Operations→Restore Backup. . .**.
The Restore Backup window displays.

2. In the **I want to restore** drop-down list, select from one of the following options:

- **Database from a PC**: This is the selected default option. Only the system database, not any system data, will be restored from the backup file as was done in releases prior to 6.0.

- **Database from a PC and voice data elsewhere**: Select this option if Voice Data is saved at an alternate location (such as a USB thumb drive). Enter the voice data information separately. Only the system database is restored from the backup file and not any system data. See "Saving and Restoring Voice Processing Databases" on **page 1001** for additional details.

- **Everything from a PC**: This option will perform a bundled restore operation where all available system data (Voice, HTML Apps, FB-MOH Files) will be retrieved from the backup file.

NOTE	Database Programming will use the Web Listening Port defined in the System Connection to save the additional data. If the Web Listening Port is not open, a warning message dispalys.

3. In the **Restore from** field, enter the location from where the backup will be restored, or click the ellipses at the end of the field and navigate to the location.

4. Click **Start**.

> **NOTE** | If the data is encrypted, it will be necessay to enter the username and password in the accompanying dialog. If an incorrect username or password is entered, the restore must be cancelled.

The "Restore Summary" will list what will be restored once selections are made.

> **NOTE** | If the data is encrypted, DB Programming will try to use the username/password to decrypt the data after you click **Start**, but if this combination does not work, a dialog will be presented asking for the correct username/password. If the correct username/password is not entered, the restore must be cancelled.

During the restore process, the following items are verified:

- *If the number of multi-protocol phones in the database being restored exceeds the number indicated in the license, an error message is displayed, and the database is not restored.*

- *If the features enabled in the database being restored do not match the features available in the software license,* an error message is displayed, and the system is inoperative.

- *If the database being restored contains Voice Processor information and the system does not have an operating Voice Processor,* you will see a warning message and no Voice Processor information will be loaded during the restore operation.

- *If restoring a database in offline mode,* you may receive an error message that displays, "Database is incompatible," due to the differences in system types. Use the **Change Selection** method in the System Administration & Diagnostics application when doing database restores in an offline session. This will load any database over an existing offline session database that was loaded previously without any problems.

When a database is restored, the system checks the following information against the currently loaded license:

- System type

- Number of devices

- Features

If one or more of these items in the database differ from what is programmed in the license, an error message is displayed, and the database is not restored. If the license is valid for the database to be restored, the system checks the current IP settings for the server. If any differences are detected, a screen similar to the following one displays.
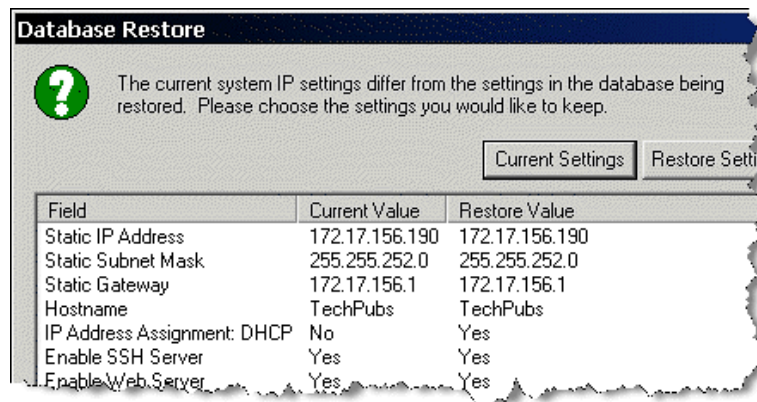
This screen displays the current IP settings (Current Value column) and the settings detected in the database to be restored (Restore Value). If you click **Current Settings**, the current settings, as shown in the Current Value column, are retained. Any other settings will be restored from the selected database. If you click **Restore Settings**, the IP settings are automatically changed to those shown in the Restore Value column, overwriting the current settings.

If, however, you defaulted the database or uploaded new software, the system considers the static database as "corrupt." When the system is powered up, a backup database is not detected, and the IP settings programmed in the restored database are automatically used.

BACKING UP THE DATABASE

A backup copy of the system database is automatically stored on the compact flash-type Mitel memory card located on the back of the HX chassis. This backup remains available, even if the system experiences a power failure or if the battery backup loses power. If the system is powered up and a corrupt database file is detected, the backup file in flash memory or on the compact flash (CF) drive is restored. You can program an automatic backup as described below.

NOTE The Voice Processor database is *not* stored in flash memory.

This backup database is designed to recover in the event of a total power failure—it is *not* designed to replace the static database. Changes made to the static database are not reflected in the backup database until the database is backed up. For example, if you default the database, the static database is defaulted, not the backup database.

If the static database is corrupted, the system attempts to restore the backup database. If the system detects problems with the backup database, it may default the static database instead of restoring the backup database. The database will be defaulted if one of the following occurs:

- There is no backup database available.

- The system was saving the backup database when the power down occurred.

- The system repeatedly attempted to restore a backup database but was interrupted by power ups. The system will default the database after it has exceeded the number of restore attempts. If this occurs, the backup database remains in flash memory for troubleshooting purposes.

If desired, you can force DB Programming to save the backup file immediately, as described in "Save Backup" on **page 60**. To set Backup Database Save parameters, see **page 65**.

DATABASE BACKUP OPTIONS

Database Backup options are located in the System folder, as shown in the figure below. For information about voice processing database maintenance, see "Saving and Restoring Voice Processing Databases" on .

| NOTE | The voice processor database is not stored in flash memory. |
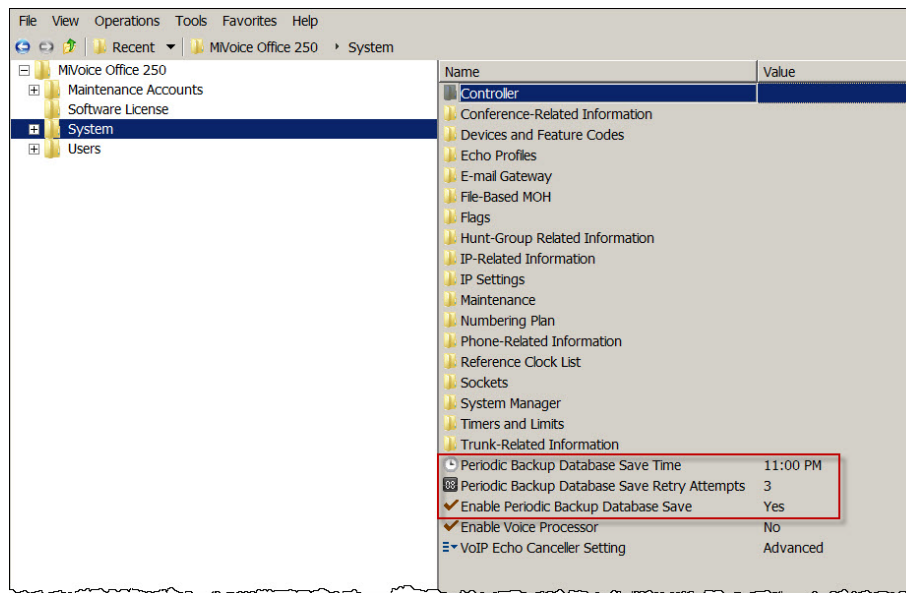
The following are database backup options:

- **Periodic Backup Database Save Time**: The time of day when a scheduled database backup occurs. This must be at least 15 minutes after the Scheduled Reset Time; otherwise, the backup may not occur. If you enter a time that is less than 15 minutes after the Scheduled Reset Time, a message appears, requesting a new value. By default, this is 11:00 P.M.

| NOTE | If Daylight Saving Time is enabled, Mitel recommends that you do *not* program the backup to occur at 2:00 PM. If you do, the backup may not occur on the day that the time changes. |

- **Periodic Backup Database Save Retry Attempts**: The number of times that DB Programming attempts to save the database. If the database is not saved before the system retries the number of times specified in this field, the database is not saved until the next day. The range is 0–10; the default is 3.

- **Enable Periodic Backup Database Save**: Enable this option to save the database at the time indicated in the Periodic Backup Database Save Time option. This option must be enabled to backup the database at the scheduled time. By default, the option is enabled.

To disable the Enable Periodic Backup Database Save option:

1. Select **System**. Backup options are shown in the right pane.

2. In the **Value** column, clear the check box. The field changes to **No**. To enable the option, select the check box.

3. Click out of the field or press **ENTER** to save the change.

NOTE	If the system detects that the database has not changed since the last backup, the save is not performed.

SCHEDULED BACKUPS

Refer to the *MiVoice Office 250 System Administration & Diagnostics Guide* for information about scheduled backups.

DEFAULT DATABASE

You can default the database to return the call processing and voice processor databases to default values. When you select **Default Database**, a window appears, as shown below, that warns that defaulting the database overwrites the current database and prompts you to continue.

NOTE	Defaulting the database ends the programming session and drops all calls. It also causes voice processing to stop. Voice processing restarts after the database default operation is complete.

From the Mitel DB Programming menu, select **Operations**→**Default Database**. The dialog box at right appears.

The default option is set to Default System Database ONLY. The second option defaults both the System database and the Backup database if one has been saved. Either option causes DB Programming to close, as expected.
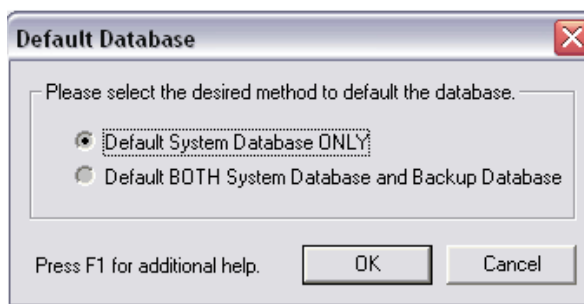
To default both System Database and Backup Database:

NOTE	Mitel recommends that you do *not* delete the "Backup Database." The "Backup Database" is created as a safety net in rare situations where the system loses its primary database.

1. Select **Default BOTH System Database and Backup Database**.

2. Click **OK**. System Database and the Backup Database are defaulted, and the application shuts down.

If you are programming in offline mode when you select **Default Database**, the following warning message appears.
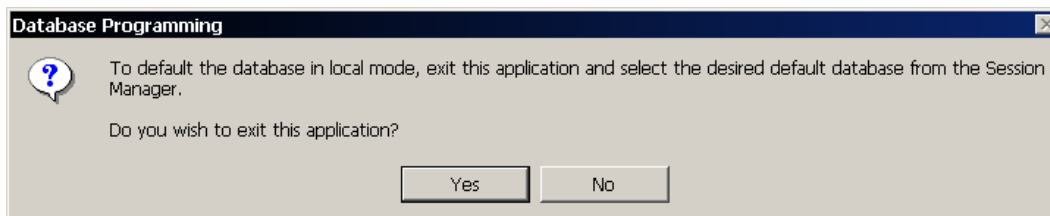
You can also default the database using the LCD panel. Refer to the *MiVoice Office 250 Installation and Adminstration Manual*, for more information.

SYSTEM ERROR INFORMATION

For system troubleshooting, you can freeze and unfreeze the system history queue and save error information.

NOTE	The history and Message Print queues are stored in RAM, not battery-backed RAM. If the system loses power, the queue information is lost. Retrieve history and Message Print information before powering down when troubleshooting the system. Queue files, however, are not affected by minor or major resets

To retrieve history and Message Print queue blocks:

1. Launch System Administration & Diagnostics, and then start a DB Programming session.

2. From the DB Programming menu bar, select Operations – **Error Information**. The following dialog box appears.
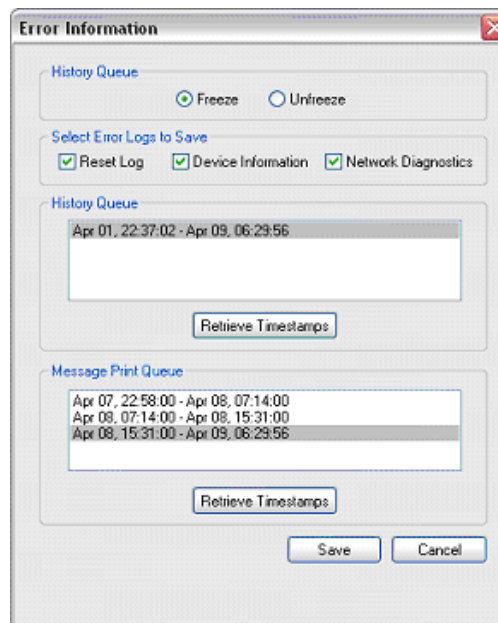
When you select Network Diagnostics option before you complete a system freeze, the freeze includes a Network Diagnostics Log file with an `.ndl` extension. This log file captures settings call processing, IP resource, and IP settings, as well as insufficient bandwidth alarms. This information can be obtained using System Administration & Diagnostics. For more information, refer to the *MiVoice Office 250 System Administration & Diagnostics Guide*.

3. If the system is not already frozen, select **Freeze**. Otherwise, continue to the next step.

4. Click **Retrieve Timestamps** to view the timestamps associated with the history and Message Print queues. The blocks are listed based on the time intervals.

5. Select the timestamps that you want to save (you can use the SHIFT and CTRL key to select more than one item).

6. Click **Save**. The standard Windows browse screen appears.

7. Select the destination for the files (maximum 65 characters, including the freeze file names), and then click **OK**.

SYSTEM MAINTENANCE OPTIONS

System maintenance options include the following:

- Call Costs on **page 68**

- Freeze Zones on **page 71**

- Immediate System Resets on **page 73**

- Message Print on **page 77**

- Station Message Detail Recording on **page 79**

System Maintenance options are under System – Maintenance, as shown in the figure below.
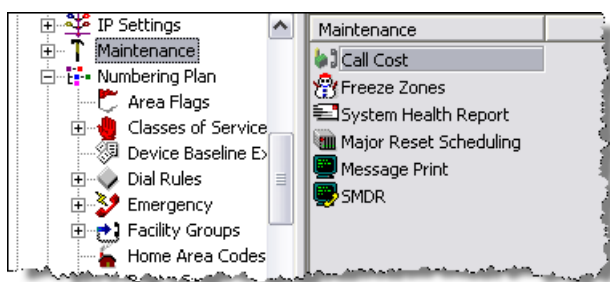
CALL COSTS

This section contains the following information:

- **Feature Description** below

- Programming Call Costs on **page 70**

FEATURE DESCRIPTION

NOTE	The system Call Cost Accounting feature provides a *cost estimate* that is applied to the various classes of calls. Due to the wide variation in charges among network carriers, the system's call cost calculation cannot be used as a prediction of actual charges. This feature can only be used as a management tool to *estimate* call costs.

The Call Cost Accounting feature estimates the cost of outgoing and incoming calls, displays it on the phones, and prints it in the SMDR records. The cost is based on the type of call, telephone number dialed, the elapsed time of the call, the day of the week, and the time of day. A table in the database supplies the rates for all types of calls, including multiplicative factors for evening and weekend rate changes on outgoing calls. The equation for calculating call cost is:

Daytime Rate X Multiplicative Factor X Connect Time

The multiplicative factor adjusts the daytime per-minute call cost for evening and weekend rates of outgoing calls. For example, the evening call cost multiplier is 0.65 if calls are 35 percent less expensive after 5:00PM. The daytime rate (D), evening (E) multiplicative factor, and night/weekend (N/W) multiplicative factors are used on the following schedule:

	Sun	Mon	Tue	Wed	Thu	Fri	Sat
8 AM TO 5 PM	N/W	D	D	D	D	D	N/W
5 PM TO 11PM	E	E	E	E	E	E	N/W
11PM TO 8 AM	N/W	N/W	N/W	N/W	N/W	N/W	N/W

If call cost is set to zero, call cost will not display during the call and the SMDR record will show $00.00.

In a network setting, the call cost shown on the display and SMDR output use the factors and rates for the node on which the trunk resides. In other words, if a phone on Node 1 dials what is considered a long-distance number on Node 1, but the call is routed to Node 2 where the number is considered local, the phone will use the local call cost rate from Node 2. The call rate used for calls between nodes will be based on the Network call rate.

All outgoing calls using a trunk that is not subject to toll restriction are classified as one of the following call types for call cost calculation (call cost type is programmed in the database).

- Free (000 or FOC)

- Local (LOC)

- Toll Local (TLC)

- Toll Long Distance (TLD or NAT)

- Operator (OP)

- International (INT)

- Operator and International (O/I)

- DISA (DSA)

- Conference (CNF)

- DID/DNIS (non-DISA) (DID or DI)

- Incoming (IN)

- Network (NET)

When a trunk that is subject to toll restriction is used, call cost type is determined according to the digits dialed, as follows:

- **Local**: The following calls are classified as local calls:
    - Calls to N11 or 1N11 (where N is 2–9), except 411 or 1411
    - Any call to a toll-free area code
    - All 7-digit or 10-digit calls within a local area code to office codes that are allowed in User Group 1

- Any call that does not begin with 1, and does not fall into any of the other call cost categories

- **Toll Local***:* The following calls are classified as toll local calls:

  - Any calls to 411 or 1411

  - All 7-digit or 10-digit calls within a local area code to office codes that are restricted in User Group 1

  - Any call that begins with 1, that does not fall into any of the other call cost categories

- **Toll Long Distance***:* Any call to an area code other than a local area code is classified as a toll long distance call.

- **Incoming***:* Any call, except DID/DNIS calls, that rings into the system and is answered is classified as an incoming call.

- **Network***:* Any call placed to or received from a network node is classified as a network call.

- **Free***:* Any call within a PBX (a trunk access code is not dialed) is a free call.

- **Operator***:* Any call starting with 0 or containing only 0 is classified as operator (0, 0+).

- **International***:* Any call starting with 01 is classified as international (01+, 011+).

PROGRAMMING CALL COSTS

This section provides information for programming Call Costs.

Selecting a Call Cost Rate

To select a Call Cost a rate:

1. Select System – Maintenance – **Call Cost**. Call Cost options are shown in the right pane.

2. In the **Rate/Factor** column, select the current value, and then type the new value in the box.

3. Click out of the field or press **ENTER** to save the change.

Calculating Call Costs

You can use the following procedure to calculate call costs.

To calculate call costs:

1. Determine the daytime rates, in dollars [pounds in Europe] per minute, for the following types of calls (all default values are 0.00). Use several service provider bills from months with typical usage to calculate the average cost per minute of each type of call. Record the charges in dollars [pounds] and cents [pence] from 00.00–99.99. You may need to adjust the calculations later to more accurately estimate actual call costs. (This program is to be used as an estimate only.

2. Determine the multiplication factors that adjust the daytime (peak) per-minute call cost for evening (standard) and weekend (cheap) rates.

   For U.S. systems: The multiplicative factor adjusts the daytime per-minute call cost for evening and weekend rates of outgoing calls. For example, the evening call cost multiplier

is 0.65 if calls are 35% less expensive after 5:00PM. The evening (E) multiplicative factor and night/weekend (N/W) multiplicative factors are used on the following schedule:

	SUN	MON	TUE	WED	THU	FRI	SAT
8 AM TO 5 PM	N/W	No Multiplication Factors Apply					N/W
5 PM TO 11PM	E	E	E	E	E	E	N/W
11PM TO 8 AM	N/W	N/W	N/W	N/W	N/W	N/W	N/W

For European systems: The multiplicative factor adjusts the peak per-minute call cost for standard and cheap rates of outgoing calls. For example, the call cost multiplier is 0.65 if calls are 35% less expensive after 6:00PM. The standard (S) and cheap (C) multiplicative factors are used on the following schedule:

	SUN	MON	TUE	WED	THU	FRI	SAT
9 AM TO 1 PM	C	No Multiplication Factors Apply					C
1 PM TO 6 PM	C	S	S	S	S	S	C
6 PM TO 9 AM	C	C	C	C	C	C	C

FREEZE ZONES

You can freeze the system to "lock" the current state of the system fault history queue, which is a sequential list of all system commands and inputs. You can use this list, when decoded, to determine a series of events which may have resulted in an error. Unfreezing the system "unlocks" the current state of the history queue, and resumes event collection.

Using freeze zones, you can determine which nodes in the network are frozen during each freeze request. There can be up to 10 freeze zones in the database.

PROGRAMMING FREEZE ZONES

To program a freeze zone:

1. Select System – Maintenance – **Freeze Zones**. Freeze Zone options are shown in the right pane.

2. *Optional.* In the **Description** column, type a name for the freeze zone in the box.

ADDING NODES TO FREEZE ZONES

Double-click an individual zone to view nodes, if any, assigned to the freeze zone.

To add nodes to the freeze zone:

1. Select System – Maintenance – **Freeze Zones**. Freeze Zone options are shown in the right pane.

2. Use one of the following methods:

Method A

   a. In the **Value** column, select the current value, and then type the new value in the box.

   b. Press **ENTER**. A screen appears displaying what is associated with the number entered.

   c. Click **OK**. The new number appears in the field.

Method B

   a. Right-click anywhere in the right pane, and then select **Add To List**. A window appears prompting for the device type to include.

   b. Select the node types (you can use the SHIFT or CTRL key to select more than one item), and then click **Next**. The items with details appear. To view items in a list only, click **List**.

   c. Select the appropriate nodes, then select **Add Items**.

   d. When you have added all the necessary nodes, click **Finish**. The selections appear in the list. To view programming options, double-click the extension number.

DELETING NODES FROM FREEZE ZONES

To delete nodes from a freeze zone:

Select one or more nodes from the list, right-click, and then select **Remove Selected Items**.

SYSTEM RESETS

This section describes system resets and how to reset the system on demand or to schedule a reset. The system *does not* require a reset as a result of configuration in a DB Programming session.

> **NOTICE**
>
> **Possible Service Interruption.** A system reset terminates all calls in progress. Schedule resets to occur after normal business hours.

This section contains the following information:

- **Immediate System Resets** below
- **Call Processing Resets** below
- Major Reset Scheduling on **page 73**
- Message Print on **page 77**
- Station Message Detail Recording on **page 79**

IMMEDIATE SYSTEM RESETS

You manually reset the system. You can also use the LCD panel on the Base Server to immediately reset the system. For more information, refer to the Installation chapter in the *MiVoice Office 250 Installation and Adminstration Manual*.

To perform an immediate system reset:

1. From the DB Programming menu bar, select Operations – **Reset System**.

2. Click **OK** to reset the system and end the programming session.

CALL PROCESSING RESETS

To reset Call Processing:

1. From the DB Programming menu bar, select Operations – **Reset Call Processing Application**.

2. Click **OK** to reset call processing and end the programming session.

MAJOR RESET SCHEDULING

A major system reset terminates all calls and ends the programming session. However, a delayed system reset checks system activity at the scheduled time and waits for the system to become idle and any active programming session to end before allowing the reset to occur.

A Call Processing reset affects call processing and all related applications. To avoid Call Processing issues, you can schedule a delayed reset, as described in the following sections.

NOTE	If Daylight Saving Time is enabled, Mitel recommends that you do not schedule resets to occur at 2:00 AM. If you do, the system may not perform the reset when the time changes.
	Make sure the Scheduled Reset Time value (it defaults to 12:01AM) is AFTER the Periodic Backup Database Save Time (it defaults to 11:00PM); otherwise any Database changes since the previous backup database save will be lost.

SYSTEM REQUIRES RESET

(Read Only) Indicates if Call Processing requires a reset. If the field shows *Yes*, the system resets at the time indicated in the System Delayed Major Reset field. If the field shows *No*, a delayed major reset does not occur, even if one is scheduled.

SCHEDULED RESET TIME

If the System Requires Reset option is turned on, this field determines the default time for *scheduled* system resets. Scheduled major resets can be scheduled when system administrators program the database or when Reset System is selected from the Operations menu.

To program the default time for delayed system resets:

1. Select System – Maintenance – Major Reset Scheduling – **Scheduled Reset Time**.

2. In the **Value** column, select the time for the scheduled reset (AM/PM). The default value is 12:01 AM.

3. Click out of the field or press **ENTER** to save the change.

FORCE RESET IF NOT IDLE

Normally, the system does not perform a major reset if there are any active calls. However, if this option is turned on, the system forces a major reset at the specified time, as programmed in the previous section. A major reset causes all active calls on the system to be dropped. The option should be used only on systems which are busy 24 hours each day and, therefore, do not have a consistent time when all resources are idle and a normal delayed major reset can be performed. This option affects any request to perform a major reset, whether it be by the system itself for resource reconciliation, or requested through DB Programming.

NOTE	When enabled, this option drops all active calls at the specified time, should a major reset be necessary. This does not happen every day, but it does happen occasionally. Be aware of this so you do not mistake the reset for a system failure.

To enable the Force Reset If Not Idle option:

1. Select System – Maintenance – Major Reset Scheduling – **Force Reset If Not Idle**.

2. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

3. Press **ENTER** or select another field to save the change.

DAYS OF THE WEEK

Select the days of the week on which you want automatic resets to occur. Resets occur on the days of the week that are selected, provided the **Always Reset On Days Of Week** option is turned on. By default, all days of the week are turned off.

To select days:

1. Select System – Maintenance – Major Reset Scheduling – *<day>*.

2. In the **Value** column, select the check box. The field changes to **Yes**. To deselect the day, clear the check box.

3. Press **ENTER** or select another field to save the change.

ALWAYS RESET ON DAYS OF WEEK

Set this option to *Yes* to have resets occur on the specified days of the week. If this option is disabled, resets do not occur on the specified days. By default, this option is set to No. If a red "X" appears next to **Always Reset On Days Of Week**, you have not selected any of the days of the week. You must have at least one day of the week selected; otherwise, resets do not occur. When finished, the system resets as programmed.

To enable Always Reset on Days of Week:

1. Select System – Maintenance – Major Reset Scheduling.

2. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

3. Press **ENTER** or select another field to save the change.

REBOOT SYSTEM

The Reboot System flag causes scheduled delayed resets to reboot the entire system. For systems equipped with a PS-1/vPS-1, this option reboots the PS-1/vPS-1 and the Base Server. It is set to No by default.

RESET SYSTEM DIALOG BOX

When you make a change that requires a reset, the Reset System dialog box appears when you exit DB Programming.
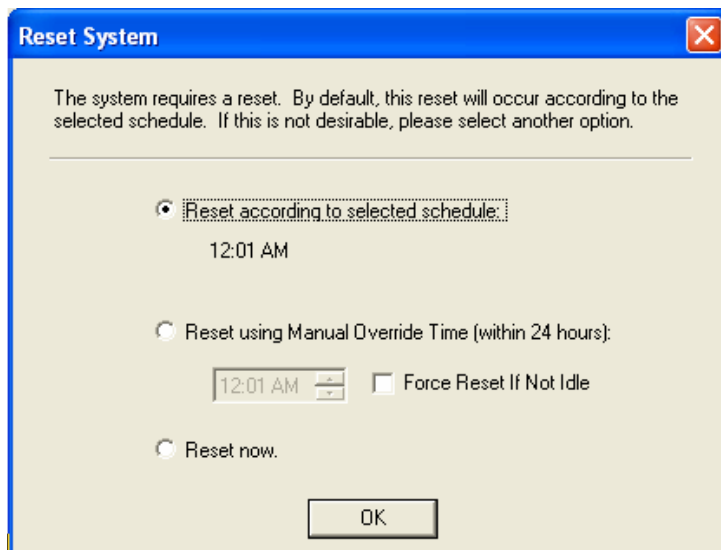
1. Select one of the following options:

    *   **Reset according to selected schedule**: Resets the system according to the schedule that the user selected in the Major Reset Scheduling folder (System\Maintenance). The reset schedule that is currently selected appears here.

    *   **Reset using Manual Override Time (within 24 hours)**:

NOTE	The Reset using Manual Override Time (within 24 hours) value must be at least 15 minutes before the programmed Periodic Backup Database Save Time value (under System); otherwise, the backup may not occur. If you select a value that is less than 15 minutes before the Periodic Backup Database Save Time, an error message is displayed, requesting a new value

    You can use the scroll box to change the scheduled reset time to a different time (within 24 hours). For example, if the system is programmed to reset on Wednesdays at 11:30 PM, and you schedule a manual override time for 10:00 PM on a Wednesday, the system resets at 10:00 PM and then again at 11:30 PM. By default, this is 12:01 AM. Note that the Manual Override Time is only used when the system requires a reset. This option can also be accessed from the Major Reset Scheduling folder in OLM mode.

    *   **Force Reset If Not Idle**: Turn on this option to force a reset, even if there are active calls. If turned off, the reset is not performed until the system is idle. By default, this is set to No. This option can also be accessed from the Major Reset Scheduling folder. (This field was previously called Forced Delayed Major Reset.)

    *   **Reset now**: Performs an immediate system reset. Do not select this option if the system is currently backing up the database; otherwise, the backup is aborted.

2. Click **OK** when finished.

MESSAGE PRINT

System messages can be printed to give service personnel and Mitel engineers information about system status during troubleshooting. You may enable any combination of the error message types. The available message types are:

- **Alarm Messages**: Indicate that a minor alarm has occurred, but that general system operation was not affected.

- **Information Messages**: Provide information concerning system operation.

- **Severe Messages**: Indicate that a severe error has occurred in the system.

- **Warning Messages**: Indicate that a condition exists which may affect system performance.

- **Network Dump**: Provides information concerning network operation.

The fields you must program to set up Message Print include the following:

- **Output Port And Local Backup Port** below

- Message Print Output Active on **page 78**

- Output Device Line Width on **page 78**

- Print Options on **page 78**

For more information about Message Print, refer to the *Message Print Diagnostics Manual*.

OUTPUT PORT AND LOCAL BACKUP PORT

Each node has its own Message Print programming, Message Print output port, and Message Print output port backup. There should be a Message Print terminal at each node to monitor node and network performance and aid in troubleshooting.

- If a node Message Print output port is a node, the network sends Message Print records to the specified node.

- You cannot select a node as the backup Message Print output port.

- If Message Print output programming forms a loop, the system sends the output to the node backup Message Print port. For example, if the Message Print port on Node 1 routes to Node 2 and the Message Print port on Node 2 routes to Node 1, the configuration causes an infinite loop. Message Print reports for Node 1 would be printed to the backup serial port on Node 2 and vice versa.

To select ports for the Message Print reports, use one of the following methods:

NOTE	Because a serial interface is not available on the MiVoice Office 250, the output to Message Print is sent over IP to a remote node.
	This remote node can only be an Axxess system, and supported networking with Axxess systems ended with Release 6.0.

Method A

1. Select System – Maintenance – Message Print – **Output Port** or **Local Backup Port**.

2. In the **Value** column, select the current value, and then type the new value in box. The port number must be a port on a remote node.

3. Click out of the field or press **ENTER**. A screen appears showing what is associated with the number entered.

4. Click **OK**. The new number appears in the field.

Method B

1. Select System – Maintenance – Message Print – **Output Port** or **Local Backup Port**.

2. Right-click the existing port. An option box appears.

3. Select **Change Port**. A window appears prompting for the device type to include.

4. Select **None** or **Remote Node**, and then click **Next**. The list of ports or nodes appears. To view items in a list only, click **List**.

5. Select the desired port, and then click **Finish**. The selection appears in the appropriate port field.

MESSAGE PRINT OUTPUT ACTIVE

When enabled, activates the error/message reporting feature. This option is enabled by default.

To disable the option:

1. Select System – Maintenance – Message Print – **Message Print Output Active**.

2. In the **Value** column, clear the check box. The field changes to **No**. To enable the option, select the check box.

3. Click out of the field or press **ENTER** to save the change.

OUTPUT DEVICE LINE WIDTH

Indicates whether the output device has 64, 80, or 132 character columns.

To set the output device line width:

1. Select System – Maintenance – Message Print – **Output Device Line Width**.

2. Select the current value, and then scroll to the desired column width.

3. Click out of the field or press **ENTER** to save the change.

PRINT OPTIONS

Determine the types of error messages to be included in the error report. By default, the following options are included in the error report (set to Yes):

•   **Print Alarm Messages**: Indicate that a minor alarm has occurred, but that general system operation was not affected.

•   **Print Information Messages**: Provide information concerning system operation.

•   **Print Severe Messages**: Indicate that a severe error has occurred in the system.

- **Print Warning Messages**: Indicate that a condition exists which may affect system performance.

- **Print Network Dump**: Provide information concerning network operation.

To disable a print option:

1. Select System – Maintenance – Message Print – *<print option>*.

2. In the **Value** column, clear the check box. The field changes to **No**. To enable the option, select the check box.

3. Click out of the field or press **ENTER** to save the change.

STATION MESSAGE DETAIL RECORDING

This section contains the following information:

- **Feature Description** below

- Programming SMDR on **page 82**

- Accessing the SMDR Socket Directly on **page 87**

FEATURE DESCRIPTION

Station Message Detail Recording (SMDR) is a system feature that provides a detailed record of outgoing and incoming calls. The system records only valid calls. Outgoing answered calls become valid when, depending on system programming, the Valid Call timer expires or polarity reversal is detected. Outgoing Unanswered calls never become valid. Outgoing calls become valid immediately if placed on hold or transferred. Incoming calls are always valid immediately. Phone call data can be retrieved into a IP-compatible SMDR report-generating device.

You can also send SMDR information to the System Manager server. Defining sockets for either output is programmed through the Administrative Web Portal (see "Administrative Web Portal" on **page 1147**), but only one output port for SMDR can be active at any time. In a network, each node has its own SMDR programming, SMDR output port, and SMDR output port backup. You can enable or disable network call records on each node. When network calls are enabled, the following applies:

- The system generates SMDR records for outgoing calls on the node where each trunk used resides. For example, if a caller on Node 1 places a call using a trunk group on Node 2, the SMDR report for Node 1 shows the outgoing call to Node 2 and the SMDR reports on Node 2 shows the incoming call from Node 1 and the outgoing call on the trunk group.

- The call record type for network SMDR records is NET.

- If a node's SMDR output port is a node, the network sends SMDR records to the specified node.

- You cannot select a node as the SMDR output port backup.

- If SMDR output programming forms a loop, the system will send the SMDR output to the node's backup SMDR port. For example, if the SMDR port on Node 1 routes to Node 2 and the SMDR port on Node 2 routes to Node 1, the configuration will cause an infinite loop of SMDR routing. SMDR reports for Node 1 would be printed to the backup serial port on Node 2 and vice versa.

Selectable SMDR options include any combination of the calls listed on **page 85**. For a definition of call cost types see "Call Costs" on **page 68**:

The system administrator can determine whether absorbed digits, equal access digits, and/or toll field digits appear in the SMDR printout. As an example, assume the following number was dialed: 89 (other system's trunk access code) + 10288 (equal access code) + 1 (toll field) + 602 (area code) + 961-9000 (seven-digit number). The following programming options can be used (see **page 86** for a definition of suppress digit options):

- **Suppress Absorbed Digits**

- **Suppress Equal Access Digits**

- **Suppress Toll Digits**

Any combination of the above can be used. If all three fields are suppressed, only 602-961-9000 will print. The system administrator can also suppress or allow call information in the SMDR report "Dialed Digits" field. The following options are available:

- **Suppress Outside Party Number**

- **Suppress Trunk Number**

The system administrator can determine which equipped phone(s) and/or trunks will be included in the report, and whether off-node devices will be included in reports. However, for incoming calls, DISA calls, conference calls, and/or ring-in diagnostics, all calls are recorded even when they involve phones not in the programmed phone list.

SMDR can be programmed to record the elapsed time of calls in seconds (S=XXXXXX) or hours and minutes (HH:MM). If programmed to record elapsed time in seconds, the ELAPSED TIME field will show "S=XXXXXX" (XXXXXX represents the number of seconds) for calls up to 999999 seconds long. For calls lasting longer than 999999 seconds, ELAPSED TIME will show "HH:MM" (hours and minutes rounded up to the nearest minute).

SMDR Output Record

The SMDR output record is printed as shown in the following example. A page heading (with the day, date, month, year and column headings) is generated just after midnight to show the change in date. A header is also printed after output from another system source (alarm, informative message, and so forth), using the same output device, interrupted the SMDR output.

Figure 5: SMDR Output Example

```
26.05.2015
TYP   EXT#  TRUNK      DIALED DIGITS      START    ELAPSED    COST      ACCOUNT CODE
XX    XXXX  XXXXXXX  <28 characters>      HH:MM    HH:MM:SS   $XX.XX    XXXXXXXXXX    *
```

In the upper left of the beginning of the SMDR Output, the date for which the data was collected is shown.

TYP

The call type (see **page 69**).

EXT

The extension number (XXXXX) of the last phone to handle the call is shown. For a CO-to-CO call, this field shows the extension number of the second trunk involved. For a conference call, it shows the phone that initiated or answered the call. For an unanswered ring-in, it shows *****. An outgoing call that has been initiated by another trunk shows the initiating trunk's number.

TRUNK

The extension number of the trunk used during the call. For an IP network call (using the networking IPRA), this field shows the extension number of the IP connection used for the call.

DIALED DIGITS

For an outgoing call: The first 28 digits of the telephone number are shown (if ARS was used to place the call, the modified number, not the dialed digits, are shown). A ">" at the end of the number indicates that more than 28 digits were dialed. Some digits may be suppressed (see the previous page for an explanation). For a conference call, this field shows the phone that brought the trunk into the conference unless the conference ends as a call with only one phone and one trunk or if all conference parties are put on Individual Hold, in which case the field shows the last party to handle the call.

For an incoming call: This field is determined by the service type of the trunk that was used for the call and whether the digits are being suppressed. If the information is not suppressed, it is included in the record, as shown in **Table 15**.

Table 15:   Unsuppressed SMDR Fields by Trunk Service Type

SERVICE TYPE	CALL RECORD SHOWS . . .	RING-IN RECORD* SHOWS . . .
DID, DNIS, or *DNIS[1] (not DISA ring-in)	Trunk DID or DNIS Information	Ring... Trunk *DID or DNIS Information*
DID, DNIS, or *DNIS* with DISA ring-in	Blank if answered; DISA digits if unanswered	DISA calls do not generate a ring-in record
Caller ID, ANI, or *ANI*	Outside Party ANI or Caller ID	Ring...*Outside Party ANI or Caller ID*
*ANI*DNIS*	*Outside Party ANI or Caller ID* **and** *Trunk DID or DNIS Information*	Ring...*Outside Party ANI or Caller ID* **and** *Trunk DID or DNIS Information*
None	Blank	Blank

1. Ring-in records appear only if ring-in diagnostic records are enabled.

START

Shows the time that the call became valid. For an unanswered ring-in, it shows the time the call began ringing. It is shown in 24-hour time (00:00–23:59).

ELAPSED

Shows the call length from the START time (above) until disconnect or the length of time an unanswered call was ringing. If the option is enabled that shows call duration in seconds, calls up to 999999 seconds long appears as S=XXXXXX (XXXXXX represents the number of seconds) and calls lasting longer than 999999 seconds appear as HH:MM (hours and minutes).

COST

Displays an estimated cost of the call ($XX.XX), based on the database information. If cost exceeds $99.99, it is printed without the decimal point ($XXXXX). If it exceeds $99999, $$$$$$ is shown instead. If there is no cost, the field is blank.

ACCOUNT CODE

Shows the standard, forced, or optional account code (up to 16 digits). An optional account code overrides standard or forced account codes. The field is blank if no account code was used.

(RESULT)

One of the following characters may be displayed at the end of record:

- *****: Indicates that the call was terminated because the system detected a disconnect signal from the trunk. That is, the outside caller hung up before the phone user on the call.the outside caller hung up

- **R**: Indicates that the call was answered by the outside party for Dynamic Extension Express, but the called party did not accept the call.

- **T**: Indicates that the trunk was involved in a Two B-Channel Transfer.

PROGRAMMING SMDR

You can program the following SMDR information:

- **Devices** below
- Output Port and Local Backup Port on **page 83**
- SMDR Output Active on **page 83**
- Output to System Manager on **page 84**
- Display Elapsed Time in Seconds on **page 84**
- Display "O/I" for Operator and International Calls on **page 84**
- Display Redirected Station on **page 85**
- Display "T" for Two B-Channel Transferred Calls on **page 85**
- Record Calls on **page 85**
- Suppress Digits Options on **page 86**

Devices

To assign the phones and trunks to be included in the SMDR output, double-click **Devices**. A list of current devices, if any, appears. You can add or delete devices as follows:

To add devices:

1. Select System – Maintenance – SMDR – **Devices**.

2. Right-click anywhere in the right pane, and then select **Add To Devices List**. A window appears prompting for the device type to include.

3. Select the device types (you can use the SHIFT or CTRL key to select more than one item), and then click **Next**. A list of devices appears. To view details, click **Details**.

4. Select the appropriate items, and then select **Add Items**. When you have added all the desired devices, click **Finish**. The selections appear in the list. To view programming options, double-click the extension number.

To delete devices:

Select the item(s) in the list, right-click, and then select **Remove Selected Items**.

Output Port and Local Backup Port

Each node has its own SMDR programming, SMDR output port, and a local (backup) SMDR output port. You can turn on/off network call records on each node. By default, the system suppresses network call records. However, when they are turned on, the following applies:

NOTE	Because a serial interface is not available on the MiVoice Office 250, the output to SMDR is sent over IP to a remote node. This remote node can only be an Axxess system, and supported networking with Axxess systems ended with Release 6.0.

• If the node SMDR output port is a node, the network sends SMDR records to the specified node.

• If SMDR output programming forms a loop, the system sends the SMDR output to the node local SMDR port. For example, if the SMDR port on Node 1 routes to Node 2 and the SMDR port on Node 2 routes to Node 1, the configuration causes an infinite loop of SMDR routing. SMDR reports for Node 1 would be printed to the local SMDR-associated IP address on Node 2 and vice versa.

To select the output port and local port for the Message Print reports:

1. Select System – Maintenance – SMDR – **Output Port** or **Local Backup Port**.

2. Right-click the existing port. An option box appears.

3. Select **Change Port**. A window appears prompting for the device type to include.

4. Select **None** or **Remote Node**, and then click **Next**. The list of ports or nodes with details appears. To view options in a list only, click **List**.

5. Select the port that you want to use, and then click **Finish**. The selection appears in the port field.

SMDR Output Active

The SMDR Output Active option activates the SMDR reporting feature. It is enabled by default.

To disable the SMDR Output Active option:

1. Select System – Maintenance – SMDR – **SMDR Output Active**.

2. In the **Value** column, clear the check box. The field changes to **No**. To enable the option, select the check box.

3. Click out of the field or press **ENTER** to save the change.

Output to System Manager

If the Output to System manager option is turned off, SMDR information is not available to System Manager users. If turned on, System Manager users can run call reports for this node.

To turn the option on:

1. Select System – Maintenance – SMDR – **Output to System Manager**.

2. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

3. Click out of the field or press **ENTER** to save the change.

NOTE	Because a serial interface is not available on the MiVoice Office 250, the output to SMDR is sent over IP to a remote node. This remote node can only be an Axxess system, and supported networking with Axxess systems ended with Release 6.0.

Display Elapsed Time in Seconds

To allow SMDR to give a more accurate representation of elapsed time, the Display Elapsed Time in Seconds option can be turned on to record the elapsed time of calls in seconds instead of minutes. For calls up to 999,999 seconds in length, the ELAPSED TIME field shows "S=XXXXXX" (XXXXXX represents the number of seconds). For calls lasting longer than 999,999 seconds, ELAPSED TIME shows HH:MM. Hours and minutes rounded up to the nearest minute.

To have call durations of fewer than 999,999 seconds displayed in seconds:

1. Select System – Maintenance – SMDR – **Display Elapsed Time in Seconds**.

2. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

3. Click out of the field or press **ENTER** to save the change.

Display "O/I" for Operator and International Calls

If turned on, operator and international calls are displayed in SMDR as one entry under the call-type abbreviation "O/I." If turned off, operator and international calls are displayed separately in SMDR: operator calls under "OP," and international calls under "INT." By default, this is turned on.

To turn off the Display "O/I" for Operator and International Calls option:

1. Select System – Maintenance – SMDR – **Display "O/I" for Operator and International Calls**.

2. In the **Value** column, select the check box. The field changes to **No**. To enable the option, clear the check box.

3. Click out of the field or press **ENTER** to save the change.

Display Redirected Station

A phone that transfers or manually forwards a call to the public network can be recorded in the SMDR report. To display redirected information for trunk-to-trunk calls, add the trunks and phones to the SMDR list. If the trunks and phones are not listed, the phones that transfer or manually forward CO calls are not recorded in the SMDR report for redirected calls. This option is disabled by default.

To enable the Display Redirected Station option:

1. Select System – Maintenance – SMDR – **Display Redirected Station**.

2. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

3. Click out of the field or press **ENTER** to save the change.

Display "T" for Two B-Channel Transferred Calls

When enabled, SMDR displays a "T" in the output when a Two B-Channel Transfer (TBCT) occurs. This option is disabled by default. For more information about TBCT, see ISDN PRI Two B-Channel Transfer on **page 268**.

To enable the TBCT option:

1. Select System – Maintenance – SMDR – **Display "T" for Two B-Channel Transferred Calls**.

2. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

3. Click out of the field or press **ENTER** to save the change.

Record Calls

The Record Calls options determine the content of the SMDR output. Options include the following:

- **Record All Incoming Calls:** Records all incoming calls that are answered, except DID/DNIS calls.

- **Record All Local Calls:** Records all calls that use the "local" call cost.

- **Record All Free Calls:** Records all calls that use the "free" call cost.

- **Record All Ring-in Diagnostics:** A ring-in message is recorded for every incoming call (whether answered or unanswered) to indicate how long it rang. All incoming calls are recorded, even those involving phones not listed in the phone list.

- **Record All Toll Local Calls:** (*U.S. only)* Records all calls that use the "toll local" call cost.

- **Record All Toll Long Distance Calls:** (*U.S. only)* Records all calls that use the "toll long distance" call cost.

- **Record All Toll (National) Calls**: (*Europe only).* Records all valid long distance toll calls.

- **Record All Operator Calls:** Records all calls that use the "operator" call cost.

- **Record All International Calls:** Records all calls that use the "international" call cost.

- **Record All DISA Calls:** Records all DISA calls.

- **Record All Conference Calls:** Records all conference calls. If all conference parties exit except one phone and one outside party, the call is no longer considered a conference and when completed it will be recorded using the appropriate call cost for that outside call.

- **Record All DID/DNIS Calls:** (*U.S. only)* Records all calls received through DID and DNIS.

- **Record All Trunk To Trunk Calls:** Records all calls made from one outside caller to another.

- **Record All Network Calls:** The system generates SMDR records (labeled "NET") for node-to-node Private Networking calls on the nodes where each trunk used resides. For example, if a caller on Node A places a call using a trunk group on Node B, the SMDR report for Node A shows a NET call to Node B and the SMDR reports on Node B shows a NET call from Node A. This option has no effect on the Record Off-Node Devices option described above.

- **Record All Off-Node Devices**: Records all calls placed to and received from off-node devices. (This flag has no effect on the Record All Network Calls flag described above.)

- **Record All Human-Answer Supervision**: When Human Answer Supervision is enabled (see "Human Answer Supervision" on **page 813**), if the user does not press # to answer a call that was routed to the public network, the SMDR record reflects that fact with a special character (R). When the user presses # to accept the call the system outputs the normal SMDR record. This option indicates whether or not the system outputs the "R" SMDR record (see "SMDR" on **page 813**).

To enable Record Calls options:

1.   Select System – Maintenance – SMDR – *<record option>*.

2.   In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

3.   Click out of the field or press **ENTER** to save the change.

Suppress Digits Options

The Suppress Digits options determine which digits, if any, are suppressed when the dialed digits are reported. To choose an option, select its current Value and place a check mark in the box. To remove an option, select remove the check mark. Options include the following:

- **Suppress Absorbed Digits**: Absorbed digits (on local or PBX lines) do not appear in the report if this option is selected. In the sample above, the absorbed digits (the other system trunk access code) would be suppressed so that only 10288-1-602-961-9000 appears. If absorbed digits are repeatable on a local line, the absorbed digits do not appear in the SMDR report, even when repeated.

- **Suppress Equal Access Digit**: (U.S. only) Equal access digits will not appear in the report if this option is selected. In the sample above, the equal access code field would be suppressed to print only 89-1-602-961- 9000.

- **Suppress Outside Party Number**: Caller information that is received through ANI or Caller ID [CLID] does not appear if this option is selected.

- **Suppress Toll Digits**: When this option is selected, toll digits do not appear in the report. In the sample above, the toll field is suppressed so that only 89-10288-602-961-9000 prints.

- **Suppress Trunk Number**: Information received through DID or DNIS [DDI] is not included in the report if this option is selected.

To enable Suppress options:

1. Select System – Maintenance – SMDR – <***suppress option***>.

2. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

3. Click out of the field or press **ENTER** to save the change.

ACCESSING THE SMDR SOCKET DIRECTLY

With the MiVoice Office 250, you can directly access the SMDR socket.

To access the SMDR socket:

1. Open a TCP/IP connection to the MiVoice Office 250 on port 4000.

2. Send a login message to the MiVoice Office 250.

SMDR Login Message

To establish a connection, the client must first establish a TCP/IP socket connection. After a valid connection has been established, which is independent of any user data actually being sent or delivered, the client must send a particular login sequence as its first message to the server. This message must take the following format.

Figure 6: SMDR Login Message Format

```
<Byte_Count><Socket_Type><Password>0x00
```

This message includes the following information:

- **Byte_Count**: Indicates the byte count for the data being passed. Note that the four bytes of the byte count is included in the message, and it is in little-endian format.

- **Socket_Type**: Passed as hex 84 (0x84) to indicate an SMDR connection.

- **Password**: Can include up to 15 ASCII characters and is null-terminated. This password must match the MiVoice Office 250 SMDR password. The password is configured via Database Programming (System\Sockets).

For example, a client with the password 12345678 could use the following to log in.

Figure 7: SMDR Login Message Example

```
0x0E 0x00 0x00 0x00 0x84 0x31 0x32 0x33 0x34 0x35 0x36 0x37 0x38 0x00
```

The follow example is an SMDR record in hex format as it is sent from the MiVoice Office 250. Note that the first four bytes is the length, and the last two bytes are a carriage return (CR) and a line-feed (LF).

87

Figure 8: SMDR Record in Hex Format

```
0x52  0x00  0x00  0x00  0x54  0x4C  0x43  0x20  0x31  0x31  0x32  0x30  0x32  0x20  0x39  0x34
0x30  0x30  0x33  0x20  0x39  0x36  0x31  0x2D  0x39  0x30  0x30  0x30  0x20  0x20  0x20  0x20
0x20  0x20  0x20  0x20  0x20  0x20  0x20  0x20  0x20  0x20  0x20  0x20  0x20  0x20  0x20  0x20
0x20  0x30  0x38  0x3A  0x33  0x36  0x20  0x30  0x30  0x3A  0x30  0x30  0x3A  0x30  0x33  0x20
0x24  0x30  0x30  0x2E  0x30  0x30  0x20  0x20  0x20  0x20  0x20  0x20  0x20  0x20  0x20  0x20
0x20  0x20  0x20  0x20  0x0D  0x0A
```

The following example is the ASCII representation of the previous SMDR Record. Note that first row of bolded numbers represents the column (from 1 to 80) in which the specific SMDR field is located (e.g., the dialed digits field 961-9000 starts at column 17, which is represented by a 7).

Figure 9: SMDR Record in ASCII Format

```
12345678901234567890123456789012345678901234567890123456789012345678901234567890
TLC 11202 94003 961-9000                        08:36 00:00:03 $00.00
```

SMDR Formatting

Each SMDR record contains 80 ASCII characters followed by a carriage return (CR) and a line-feed (LF). Each field of information (for example, Type, Start Elapsed, Cost, etc.) starts in the same column number, and a space is used to separate the fields.

After the SMDR application connects to the MiVoice Office 250 or at the start of each day, the MiVoice Office 250 sends two lines of header information. Note that the first row of bolded numbers represents the column in which the specific SMDR field is located (from 1 to 80) and is not part of the SMDR output.

Figure 10: SMDR Output Format 1

```
12345678901234567890123456789012345678901234567890123456789012345678901234567890
Station Message Detailed Recording HH:MM:SS MM-DD-YYYY
```

A normal SMDR record has the following format. Note that the first row of bolded numbers represents the column in which the specific SMDR field is located (from 1 to 80) and is not part of the SMDR output.

Figure 11: SMDR Output Format 2

```
12345678901234567890123456789012345678901234567890123456789012345678901234567890
ccc eeeee ttttt dddddddddddddddddddddddd hh:mm hh:mm:ss $cc.cc aaaaaaaaaaaa*
```

A ring record has a slightly different format. Note that the first row of bolded numbers represents the column in which the specific SMDR field is located (from 1 to 80) and is not part of the SMDR output.

Figure 12: SMDR Output Ring Record

```
1234567890123456789012345678901234567890123456789012345678901234567890
***** ttttt RING........                    hh:mm hh:mm:ss
```

Below is a sample of SMDR output.

Figure 13: SMDR Output Example

```
     Station Message Detailed Recording 07:39:38 07-14-2010

TYP EXT#  TRUNK DIALED DIGITS START ELAPSED   COST   ACCOUNT CODE
LOC 11201 94003 961-9000 09:28 00:00:05 $00.00
TLD 11205 94002 1-212-555-5346 10:29 02:05:06 $12.50*
LOC 11202 94006 961-9000 10:39 01:10:03 $00.00
***** 94000 RING......480-555-9911 11:37 00:00:06 *
```

The information provided in this section is limited to the description of the SMDR socket connection to the MiVoice Office 250. For additional SMDR feature and programming information, see **page 79**.

Membership in the Mitel Solutions Alliance (MSA) at the Developer Basic or Developer Advanced member level is required to obtain Developer Support regarding MiVoice Office 250 SMDR use and troubleshooting. See **www.mitel.com/msa** for additional information or to apply online.

Chapter 4

PRIVATE NETWORKING AND SYSTEM NODES

INTRODUCTION

This chapter provides information to help you configure network settings or add system nodes to a private IP or T1/E1 PRI network.

For more information about private networks and nodes, refer to the following chapters in the *MiVoice Office 250 Installation Manual* :

- Appendix A: Private Networking

- Appendix B: Network IP Topology

For IP Networking-related troubleshooting issue, see **page 118**.

PRIVATE IP NETWORKING PROGRAMMING GUIDELINES

System software provides a networking wizard to simplify the required programming for private networking. See "Launching the Networking Wizard" on **page 99** for instructions on how to launch the wizard. For complete information about the Networking Wizard, refer to the *MiVoice Office 250 DB Programming Help*.

The Internet Protocol Resource (IPRA) application relies on a unique numbering plan to make connections across the network. You must assign each Processor Module (and if applicable, the Processor Expansion Card) in the network a unique extension that the other Processor Modules must be able to recognize. For this reason, Mitel strongly suggests that you use the P8XXY numbering convention, where *XX* is the node number and *Y* is the IPRA number. For example, if the IPRA is the second IPRA on node 1, the extension number would be P8012.

You should also complete the IP Networking Planning Sheet (see Appendix A: Private Networking in the *MiVoice Office 250 Installation Manual*) before you begin programming IP networking. The Private Networking section includes programming examples. If you do not follow a numbering convention and identify the extensions before you program the IPRAs, it will be difficult to successfully set up your network.

To program IP private networking:

1. Program the system for IP (Private Networking).

   If necessary, choose a "P6*XXY*" extension where *XX* is the node number where the IP resource resides and *Y* is the IP resource number. A default extension (P6000) is provided, but it may need to be changed if the same extension already exists. If using the Processor Expansion Card (PEC), the default extension is P6001.

NOTE	The extension you assign to each IP resource must reflect the node where the IP resource resides. If it does not, it will be difficult to establish the IP connections between IP resources on a multi-node system. This numbering convention also limits the number of networking IP resources within a private network to 1000.

2. **Program the IP Connection:** The IP resource screen contains a link to the IP resource IP connection. Whenever the system creates an IP connection, the system broadcasts the IP information to the other nodes as a database update, and the new IP connection becomes an off-node device on the other nodes (provided the remote node does not block the database change such as when a DB Programming session is active on the remote node). This does not automatically configure the IP resources for use by the network. You must manually program networking parameters locally as well as on each remote node.

3. **Program the IP Connection for the remote node:** You can also create IP connections to represent IPRAs on other nodes within the network. To see these off-node IP connection fields, you must have a node created in System\Devices and Feature Codes\Nodes.

4. Create an off-node IP connection in the IP Connections field.

5. Enter the "P6XXY" extension of the off-node IP connection. Again, you must assign a unique extension to each IP connection within the network. When creating off-node connections and other IP-related extension numbers, use a numbering plan that associates the extension to the device and the node on which it resides. For example, the first IP resource on
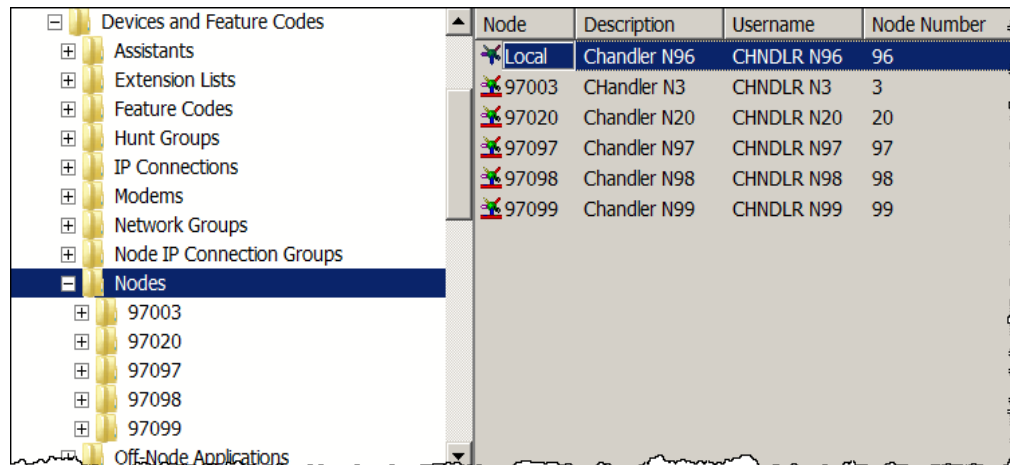
node 2 would be P6021 (P6 followed by the node number, then the IP resource number). The second IP resource would be P6022, and so forth.

6. Program a remote IP address, remote audio receive port, and remote listening port in the IP Connection screen.

7. **Program the Node IP Connection Group:** The off-node IP connection screen also contains a link to the Node IP Connection Group field to which the IP connection belongs. The system automatically creates a node IP connection group for each remote node you program. A node IP connection group corresponds to an IP network connection between the local node and a remote node. Changes to the Node IP Connection Group parameters do not take effect until the next call. Changes do not affect existing calls.

NODES

The following sections describe local and remote nodes, as shown in the figure below, and how to create and program them. You must restart DB Programming when you create a node.

You can also use the Networking Wizard to quickly add and configure new nodes. See "Launching the Networking Wizard" on **page 99** for instructions on how to launch the wizard. For complete information about the Networking Wizard, refer to the *MiVoice Office 250 DB Programming Help*.

LOCAL NODES

The local node is automatically created in DB Programming when the system is installed. For the local node, you can only assign the description, username, and node number.

REMOTE NODES

Remote nodes are Mitel Advanced Communication Platform systems, such as the MiVoice Office 250 which are connected to the local node. You must program each remote node with a node number, node trunk groups, and a search algorithm.

> **NOTE** Due to licensing changes in v4.0, an Extended Node Capacity license is no longer required to support above 63 nodes.

PROGRAMMING REMOTE NODE NUMBERS

If you change the node number, the system resets when you exit.

To change the current node number:

1. Select System – Devices and Feature Codes – Nodes – <*node*>.

2. In the **Value** column, type the new number in the box. If the number you entered conflicts with an existing node number, an error message appears. You must change the number to a valid value.

3. Click out of the field or press **ENTER** to save the change.

CHANGING REMOTE NODE INFORMATION

You can change the node description or user name.

To change the description and user name:

1. Select System – Devices and Feature Codes – Nodes – <***node***>.

2. Select the appropriate field, and then type the new information in the box. Descriptions can have up to 20 characters; Usernames can have up to 10 characters. Do not use slash (/), backslash (\), vertical slash (|), or tilde (~) characters in usernames. Do not use Control characters in descriptions or usernames.

3. Click out of the field or press **ENTER** to save the change.

DELETING A REMOTE NODE

To delete a remote node:

1. Save the database.

2. Restart the session in offline mode. You cannot remove a remote node while the session is in online mode.

3. Select System – Devices and Feature Codes – **Nodes**.

4. Right-click the remote node that you want to remove.

5. Select **Delete**.

6. Click **Yes** to restart DB Programming.

7. Restore the database to the system in online mode.

REMOTE NODE TRUNK/IP CONNECTION GROUPS

Each remote node has a list of node trunk or node IP connection groups that access other remote nodes. The node IP connection groups correspond to an IP connection between a remote node and a local node. For each node in the network, you must define the routes to every other node. For example, in a network with four nodes, you would define three routes for each node (one to each of the other three nodes). For an example of node programming, refer to the *MiVoice Office 250 Installation Manual* .

PROGRAMMING REMOTE NODE TRUNK/IP CONNECTIONS GROUPS

To program the node trunk or node IP connection groups included in this node:

1. Select System – Devices and Feature Codes – Nodes – <***node***>.

2. Double-click **Node Trunk/IP Connection Groups**.

3. Do the following:

    To add trunk or IP connection groups to the list:

    Note that this is an *ordered list*. Place the trunk or IP connection groups in the order you want them to be accessed when the hunt group (if applicable) receives calls.

- *To add to the bottom of the list:* Do not select any existing trunk or IP connection groups.

- *To add to the list above an existing trunk group:* Select the trunk or IP connection group.

a. Right-click in the right pane, and then click **Add To Node Trunk/IP Connection Groups List**. A window appears prompting for the device type to include.

b. Select **Node Trunk Group** or **Node IP Connection Group**, and then click **Next**.

c. The items with details appear. To view items in a list only, click **List**. Select the items (you can use the SHIFT or CTRL key to select more than one item), and then click **Add Items**.

d. When you have added all the node trunk groups necessary, click **Finish**. The selections appear in the list. To view programming options, double-click the extension number.

- To move a trunk or IP connection group to another location in the list:

   Drag and drop the trunk or IP connection group to the new position. Or, select the trunk or IP connection group to move and press **CTRL** + the up/down arrow to move the trunk or IP connection group up or down in the list.

DELETING NODE TRUNK GROUPS

To delete a node trunk group:

1. Select System – Devices and Feature Codes – Nodes – *<node>*.

2. Double-click **Node Trunk/IP Connection Groups**.

3. Select the node trunk or IP connection group.

4. Right-click, and then select **Remove Selected Items**. The item is removed from the list.

USING A REMOTE NODE SEARCH ALGORITHM

The search algorithm determines whether the node trunk groups are accessed in linear or distributed order:

- **Linear**: The system first attempts to route a call through the first node trunk group listed. If it is unable to route through that node trunk group, it attempts to route the call through the second node trunk group in the node. The system continues to attempt to route the call through the subsequent node trunk groups listed in the node until it successfully routes the call or exhausts all node trunk groups in the list.

- **Distributed**: The system equally distributes the first node trunk group used with each call. For example, if the system routed the first call through the first node trunk group in the node, it routes the second call through the second node trunk group in the node.

Determining the order to list the Node Trunk or IP Connection Groups in a node and when to use Linear or Distributed search type depends on your system configuration and traffic. For example, if the Node Trunk or IP Connection Group List has more than one trunk group that connects to the same node, you should use the Linear search type instead of Distributed. For more information, refer to "Appendix A: Private Networking," in the *MiVoice Office 250 Installation Manual* .

To set the search algorithm:

1. Select System – Devices and Feature Codes – Nodes – *<node>* – **Search Algorithm**.

2. In the **Value** column, select either **Linear** or **Distributed**.

3. Click out of the field or press **ENTER** to save the change.

PROGRAMMING REMOTE NODE AUDIO FOR CALLS CAMPED ONTO THIS DEVICE

The Audio for Calls Camped onto this Device field defines the audio that callers hear when camped-on to the node trunk or IP connection group. For more information about audio settings, see "Device Audio for Calls Settings" on **page 414**.

To program the Audio for Calls Camped onto this Device field:

1. Select System – Devices and Feature Codes – Nodes – *<node>* – **Audio for Calls Camped onto this Device**.

2. In the **Value** column, select the option from the list.

3. Click out of the field or press **ENTER** to save the change.

LAUNCHING THE NETWORKING WIZARD

The Networking wizard allows you to quickly configure IP networking or T1/E1 PRI networking for existing or new nodes. See "Launching the Networking Wizard" on page 99 for instructions on how to launch the wizard. For complete information about the Networking Wizard, refer to the MiVoice Office 250 DB Programming Help.

To start the IP/PRI Private Networking Wizard:

From the DB Programming menu bar, select **Tools**, and then select **Networking Wizard**. The Networking Wizard Welcome screen appears.

For complete information about the Networking Wizard, refer to the MiVoice Office 250 DB Programming Help.

IMPORTING AND EXPORTING NODE DEVICES

Perform this option after hours, when system usage is at a minimum. When another node adds a new extension or changes an existing extension, username, or description, the network broadcasts the new extension information to the other nodes in the network. Or, you can export local device information to other nodes, or import devices from the other nodes to the local node, using the Export/Import option in this menu. If there is a conflicting (duplicate) extension in a broadcast from another node, the receiving node ignores the new extension in favor of its existing extension number.

NOTE	In the default state, the Enable Voice Processor flag is enabled. If you do not have an external voice processing system connected to the MiVoice Office 250, disable this flag before attempting to import or export information over the network.

If the network is unable to export or import an extension to a node (automatically or using the Export/Import feature) because there is an active programming session on that node, the node is unable to communicate with its Voice Processor port, the node is down, or the links to the node are down, the new extension will not be added to or changed on that node. (If Message Print is enabled, error messages will indicate any unsuccessful broadcasts.) You must manually add or change the new extension in the node database or try to export or import it later. You may want to check each node to verify that their off-node device lists are programmed properly to allow access between the nodes.

Extension numbers that exist before the network is established will not be automatically broadcast to other nodes, until they are modified. They must be exported or imported using the Export/Import option or programmed manually at each of the other nodes. Also, each node can have only 8000 off-node device entries in its database. When that limit has been reached, new devices received through network broadcasts cannot be added to that node database.

To export or import devices:

1. Launch System Administration & Diagnostics, and then start a DB Programming session.

2. From the DB Programming menu bar, select Operations – **Export/Import Devices**. The following dialog box appears. You must have a programmed Node to export or import information from the node.
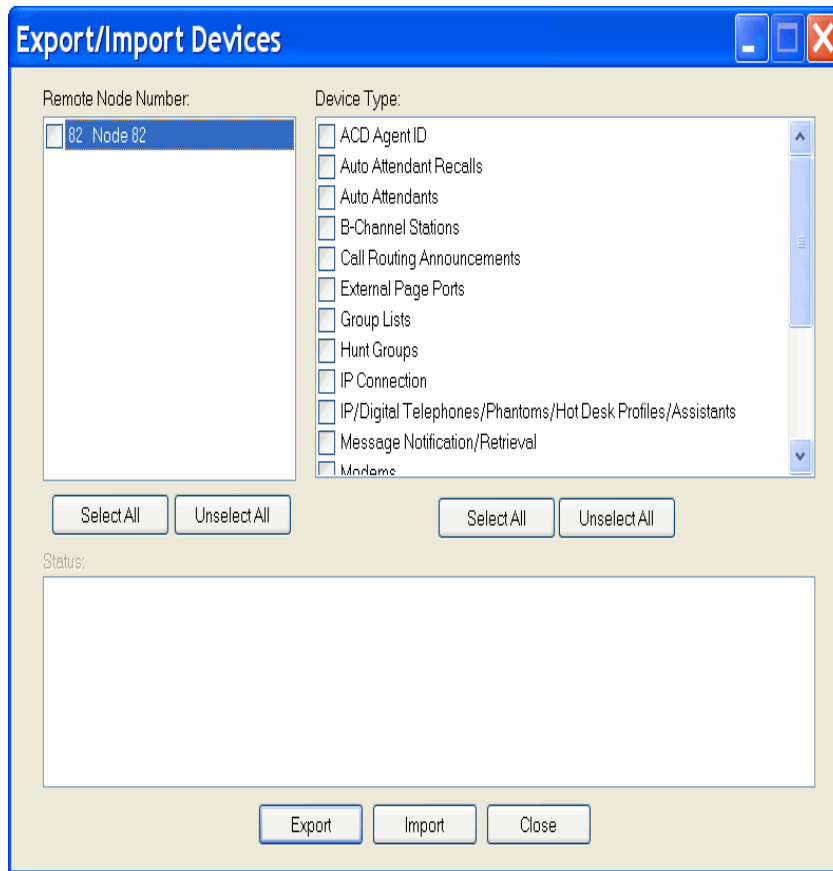
3. Select the following information, and then click **Export** or **Import**. Or, click **Close** to cancel the operation without exporting or importing any information.

    •   **Remote Node Number**: Select the node(s) you want to export the information to or import the information from, by placing checks in the appropriate boxes. Click **Select All** or **Unselect All** to select or unselect all of the nodes.

    •   **Device Type**: Select any combination of device types you want to export or import by checking the appropriate boxes. IP device information is automatically included if you import or export all phones. In addition, IP SLAs are automatically included if you import or export all single line sets. Click **Select All** or **Unselect All** to select or unselect all of the device types.

    NOTE │ ACD agents may be imported or exported the same as the other device types.

When you click to begin the export or import operation, the upper panel shows the export or import status.

If you are exporting information, the node you are exporting from is listed as "Node X: Export Source." The destination node or nodes will show the node number and the current status of the export. Once the import from a node is completed, either successfully or unsuccessfully, the import source node displays the final status of the import. The screen will show messages as explained in Table 16 on **page 102.**

Table 16:   Node Status Descriptions

STATUS	MEANING
Starting Up	Indicates that the node is attempting to initiate the export.
Creation XX%	This node is XX percent done with the creation portion of the export.
Deletion XX%	This node is XX percent done with the deletion portion of the export.
COMPLETED	This node successfully completed the export.
Waiting...	When importing, this status indicates that this node is waiting for another node to finish before it can become an import source
Error – Not Reachable	The export to the node failed because the export source node could not communicate with this node.
Error – Phone Programming	The export to the node failed because someone on the node was performing Phone Programming.
Error – Voice Processor	The export to the node failed because the link to the Voice Processor is down.
Error – DB Programming	The export to the node failed because someone on the node was performing DB Programming.

4.   After completing an import operation, close DB Programming to update the database.

If the local node cannot communicate with the remote node, the export or import will fail. Before you attempt to export or import information, verify that you can reach the remote node by dialing the extension assigned to the remote node. If you hear dial tone, you should be able to successfully export or import information. If you receive a NOT REACHABLE message, you must determine why calls are not properly being routed to the specified remote node before you can export or import information. If you camp on to a node while trying to reach the remote node, wait for the node trunk group to become available so that you can guarantee the remote node can be reached.

5.   After attempting to export or import information, check Message Print output for error messages:

- If a programming session is active on the remote node, you cannot export information to that node until the programming session has been terminated. Also, if the remote node has a Voice Processor and the link is down, the export will fail on the remote node.

- If an existing extension on the remote node conflicts with an exported extension number, you will see one of the error messages listed in the following table. The error message indicates the resulting action on the remote node. In the example used in Table 17 on **page 103**, the device information was exported from local node 1 to remote node 2.

Table 17:   Example of Exporting an Extension from One Node to Another

EXTENSION # EXPORTED FROM LOCAL NODE 1	EXISTING EXTENSION # ON REMOTE NODE 2	RESULTS ON REMOTE NODE 2
Phone 1000	Phone 10	WRN DB CP Device Info Import For ID(1:'1000') Failed: Conflict with existing extension
Phone 1000	Phone 100	WRN DB CP Device Info Import For ID(1:'1000') Failed: Conflict with existing extension
Phone 1000	Phone 1000	WRN DB CP Device Info Import For ID(1:'1000') Failed: Conflict with existing extension
Phone 1000	Off-Node Device ID(3:1000)	WRN DB CP Device Info Import For ID(1:'1000') Failed: Conflict with existing extension
Phone 1000	Unassociated Mailbox 1000	Off-Node Device ID(1:1000) created and Mailbox 1000 is associated with the off-node device
Hunt Group 2000	Unassociated Mailbox 2000	Off-Node Device ID(1:2000) created and Mailbox 2000 is associated with the off-node device
Phone 1000	Unassociated Mailbox 100	Off-Node Device ID(1:1000) created and Unassociated Mailbox is deleted
Phone 1000	Off-Node SL-Set ID (1:1000)	Off-Node Device ID(1:1000) deleted and Off-Node Device ID(1:1000) created
Page Port 91000 (99000 in Europe)	Unassociated Mailbox 91000 (or 99000)	Off-Node Device ID(1:91000 or 1:99000) created and Mailbox 91000 (or 99000) remains unassociated
Phone 1000	Off-Node Device ID (1:10XX)	WRN DB CP Device Info Import For ID(1:'1000') Failed: New extension matches existing wildcard

MODEMS

You can create off-node devices for modems on the other nodes and program individual modems on the Local node, as shown in the figure below.
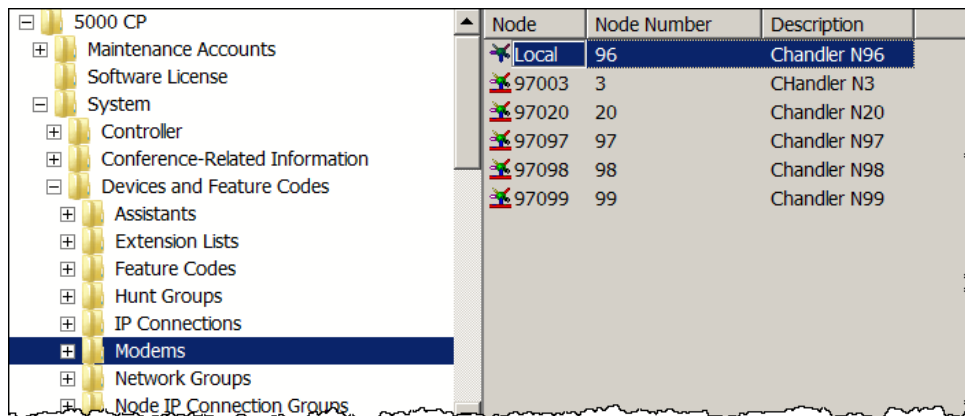
> ## NOTICE
>
> **Possible Database Corruption**. Poor line quality may cause data transmission problems when the modem connection exceeds 19200 baud. For this reason, Mitel recommends that you do *not* use the modem to restore the database. If you attempt a restore using the modem, the database may become corrupt.

OFF-NODE MODEMS

> **IMPORTANT** You cannot program off-node modems across an IP connection. Also, you must program a remote node on the system before you can create an off-node modem extension.

Off-node modems allow access to modems on other nodes. When you double-click a remote node, a list of its existing off-node page modems with extensions, descriptions, and usernames appears. You can create or delete off-node modems. After you create the modems, you can change modem extensions and enter descriptions and usernames. When programming modems, follow a universal numbering plan (for example, the extensions must be unique).

> ## NOTICE
>
> **System Instability**. Do not create or delete more than 2000 off-node devices at a time. Batch creating more than 2000 off-node devices may cause system problems.

To program off-node modem extensions:

1. Select System – Controller – Devices and Feature Codes – Modems –*<node>*.

2. Right-click in the right pane, and then select **Create Off-Node Modem**.

3. Select the starting extension and the number of modem extensions you want to create. The extension you assign to the off-node modem must match the extension programmed locally on that node. For example, if the modem extension for node **2** is **1502**, the off-node modem extension on other nodes must be **1502**.

4. For each extension, program the description (using up to 20 characters) and the user name (using up to 10 characters). After you program the off-node modem extension, you can use the off-node modem extension for the same functionality as the local modem extension.

LOCAL MODEMS

You can program local modems.

PROGRAMMING LOCAL MODEM INFORMATION

All modem extensions should have a description and a username.

To program the local modem description and user name:

1. Select System – Devices and Feature Codes – Modems – **Local**.

2. Enter the description and user name. The description can be up to 20 characters long. The user name, which appears on display phones, can have up to 10 characters. To program the names, select the desired text box and type the entry. Do not use slash (/), backslash (\), vertical slash (|), or tilde (~) characters in usernames. Do not use Control characters in descriptions or usernames.

3. Click out of the fields or press **ENTER** to save your changes.

CONFIGURING LOCAL MODEMS

To configure local modems:

1. Select System – Devices and Feature Codes – Modems – **Local**.

2. Double-click a modem extension, and then program either of the following fields:

   - **Modem Enabled**: Allows the modem to accept incoming calls, if enabled. If disabled, the modem does not accept incoming calls, and INVALID EXTENSION appears on display phones. By default, this is set to Yes. You can also use the administrator phone to reset the modem. For more information about using the administrator phone, refer to the following guides:

     - *MiVoice Office 250 Phone Administrator Guide*, part number 580.8001

     - *MiVoice Office 250 Unified Voice Messaging Administrator Guide*, part number 580.8009.

   - **Minimum Bit Rate**: Select the minimum bit rate the modem will use for its connection. The range is 9600–33600; the default value is 9600.

3. After programming the modem information, assign the modem extension as the following:

   - **Trunk Group Ring-In Destinations**: Allows remote access of the modem using a trunk call. See "Day or Night Multiple Ring-In Types" on **page 508**.

   - **Call Routing Table Ring-In Destinations**: Allows modem access using one or more specific phone numbers. See "Call Routing Tables" on **page 248**.

   - **Voice Mail Extension ID for Auto Attendant and Digit Translation**: Allows modem access through the Auto Attendant or Voice Mail extension. See "Extension IDs" on **page 920**.

SYSTEM MANAGER

System Manager is a server-based application that centralizes the management functions of the system and its peripheral products. To interface with System Manager:

- Call Processing requires a System Manager agent account for each node that will connect to the System Manager server.

- You must upload the Certification Authority (CA) certificate for System Manager. If you do not upload the CA certificate, you will not be able to access DB Programming through the System Manager Web interface.

- You must configure DB Programming to connect to System Manager.

For details about agent accounts and DB Programming requirements, refer to the *System Manager Installation and Maintenance Manual*, part no. 835.2743.

CONFIGURING THE NODE TO INTERFACE WITH SYSTEM MANAGER

To allow remote management, each system node must physically connect to the System Manager server. After a connection is established, the node is considered an "agent" to System Manager.

To configure the node to interface with System Manager:

1. Under System, double-click **System Manager**. If you have not selected a time zone, a warning message appears. You must first program a time zone before you can enable System Manager changes.

2. Configure the following options:

    - **Enable Connection**: Determines whether or not the node will attempt to connect to System Manager. Enable the option to allow the node to connect to System Manager. If disabled, the node will not connect to System Manager. By default, this option is disabled.

        a. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

        b. Click out of the field or press **ENTER** to save the change.

    - **History Capacity Limit**: Sets the size limit of the history message buffer for the Call Processing agent before messages are sent to Call Processing.

        a. In the **Value** column, click the current value.

        b. Type the new number or select the new number from the list.

        c. Click out of the field or press **ENTER** to save the change.

    - **IP Address**: Identifies the IP address of the System Manager server.

        a. In the **Value** column, click the current Value. The Edit IP Address dialog box appears.

        b. Type the correct IP address, and then click **OK**. By default, this value is 192.168.200.1.

        c. Click out of the field or press **ENTER** to save the change.

- **Password**: Identifies the password for the agent account that is programmed in System Manager. System Manager has been replaced by System Administration and Diagnostics.

> **NOTE** The **Username** and **Password** fields must match the information that is programmed in System Manager. If these fields do not match the agent account information, the node will *not* connect to System Manager.

   a. In the **Value** column, click the current value.

   b. Enter the password, up to 64 characters. This field is case-sensitive.

   c. Click out of the field or press **ENTER** to save the change.

- **Port**: Identifies the port used to connect to the System Manager server.

   a. In the **Value** column, click the current value.

   b. Type the new number. The valid range is 1025–65535; the default value is 3707.

   c. Click out of the field or press **ENTER** to save the change.

- **Require System Manager ACK**: When enabled, Call Processing sends an acknowledgement to the Call Processing agent whenever the Call Processing agent sends Call Processing a history message.

   a. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

   b. Click out of the field or press **ENTER** to save the change.

- **Send CPH to System Manager**: When enabled, the Call Processing agent sends history messages to Call Processing.

   a. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

   b. Click out of the field or press **ENTER** to save the change.

- **Username**: Indicates the user name of the agent account that is programmed in System Manager.

   a. In the **Value** column, click the current value.

   b. Enter the user name, up to 64 characters. This field is case-sensitive.

   c. Click out of the field or press **ENTER** to save the change.

- Valid CA Certificate: Read only field that shows whether or not the CA certificate is valid. If not, you must upload the certificate. See the following section.

UPLOADING THE SYSTEM MANAGER CA CERTIFICATE

If a node will connect to System Manager, you must upload the Certification Authority (CA) certificate. The certificate allows the node to use Secure Socket Layer (SSL) for secure connections.

> **NOTE** You cannot upload the CA certificate while in offline mode.

To upload the CA certificate:

1. From the DB Programming menu bar, select Operations – **System Manager CA Certifi-cate Upload**.

2. When prompted, click **Browse**, and then locate the CA certificate. The certificate (*.cer) is usually in the following directory: `C:\Program Files\Mitel\System Manager\CA`.

3. Click **Start**. The CA certificate is uploaded.

PRIVATE IP NETWORKING

This section describes how the MiVoice Office 250 handles Private IP Networking:

- **IPRA Resources** below

- **Compatibility with Existing Products** below

- Resource Allocation on **page 109**

- Resource Reservation on **page 111**

- IP Device Resource Manager on **page 111**

- Data Connections on **page 111**

- Audio Connections on **page 112**

- IP Connections and IP Connection Groups on **page 112**

- QoS on **page 112**

- Audio Connections to IP Devices on **page 112**

- DTMF Configuration on **page 113**

- Diagnostics on **page 113**

- NAT Traversal for IP Phones on **page 113**

- Automatic NAT Detection on **page 114**

IPRA RESOURCES

The Internet Protocol Resource Application (IPRA) supports both IP devices and private networking. IPRA supports up to 32 IP resources, including MGCP trunks, SIP gateway trunks, multi-protocol phones, or IP networking ports.

COMPATIBILITY WITH EXISTING PRODUCTS

The MiVoice Office 250 supports several private networking configurations with various components. Consider the component compatibility when setting up a private network. Between any two phones, there can be IPRAs or T1M or T1M-2 modules connecting T1, T1/PRI or E1/PRI spans, and IP Networking Resources. Each of these components can communicate only with like components.

For example, an IPRA on one node can communicate directly with an IPRA on another node, but not with a T1M or T1M-2 module or with an IP Networking Resource. While the IPRAs, T1M or T1M-2s, and IP Networking Resources cannot communicate directly, they can still be used in the same system multi-node network.

RESOURCE ALLOCATION

System resources for Private IP Networking are allocated in the following manner:

- **IP Networking Resources**: The system allocates IP Networking resources upon call setup. If the resources necessary to complete a call are not available, the call camps-on to the

resources and the system does not deliver the call to its destination until the resources are available.

- **Camp On**: Similar to private networking using T1/PRI or E1/PRI, the system first attempts to route a call using every programmed route. If all routes fail, the system then camps on to the first programmed route.

- **Transfers**: The system does not release IP Networking resources following a transfer, even if those resources are no longer needed for the current audio configuration.

  For example, device A on node 1 calls device B on node 2 using IP Networking Resource W on node 1 and IP Networking Resource X on node 2.
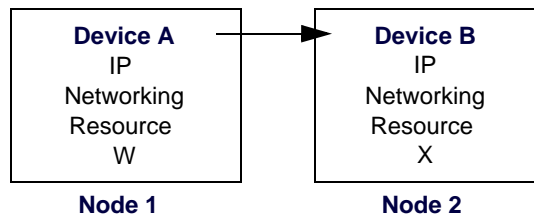
  Device B transfers the call to device C on node 3 using IP Networking Resource Y on node 2 and IP Networking Resource Z on node 3 for the transfer announcement call.
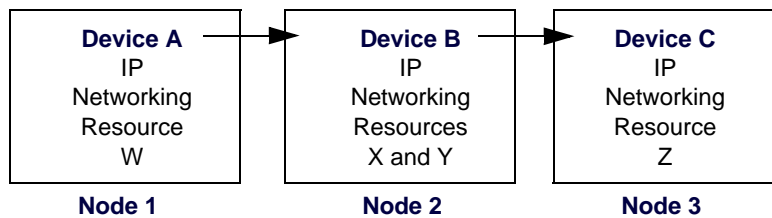
  When device B hangs up to complete the transfer, the system may optimize the route such that the system uses only IP Networking Resource W on node 1 and Resource Z on node 3 to make the audio connection. However, the system does not release Resources X and Y on node 2 for use by other calls.
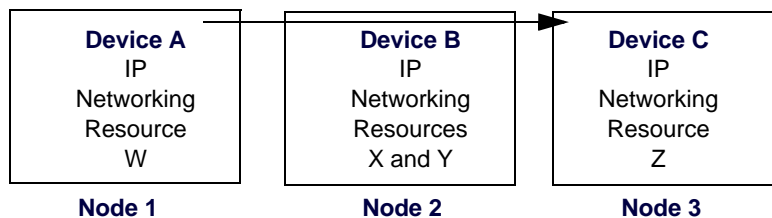
> **NOTE** Additional IP Networking Resources may be allocated if a call is transferred to multiple locations.

- **Multiple Ring-In Calls**: The system reserves IP Networking Resources for each leg of a complex call. However, once the call is answered, the system releases all Resources that are allocated for legs that no longer exist.

  For example, device A on Node 1 calls an All-Ring Hunt Group with three members on Node 2. The system reserves three IP networking resources on each node, allowing each member to receive an incoming call ring.

```
┌─────────────────┐          ┌─────────────────┐
│    Device A     │ ───────▶ │   Hunt Group    │
│       IP        │          │       IP        │
│   Networking    │          │   Networking    │
│   Resources     │          │   Resources     │
│     A, B, C     │          │     X, Y, Z     │
└─────────────────┘          └─────────────────┘
      Node 1                       Node 2
```
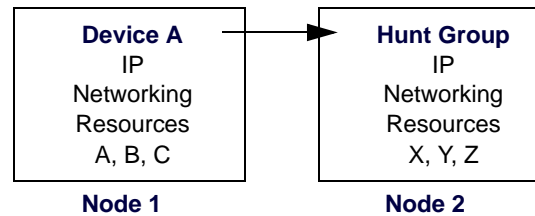
If only two IP networking resources are available on one of the nodes, the system reserves the two available IP networking resource pairs and rings only two members, applying Camp On to the third resource pair. Once a Hunt Group member answers the call, the system releases the extra resources such that only one resource on Node 1 and one resource on Node 2 are used.

```
┌─────────────────┐          ┌─────────────────┐
│    Device A     │ ───────▶ │   Hunt Group    │
│       IP        │          │       IP        │
│   Networking    │          │   Networking    │
│   Resource      │          │   Resource      │
│       A         │          │       Z         │
└─────────────────┘          └─────────────────┘
      NODE 1                       Node 2
```
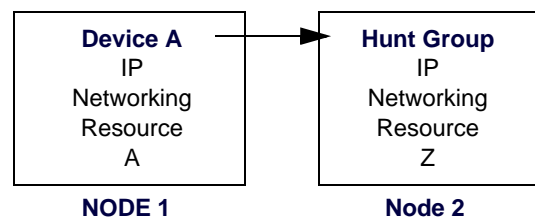
RESOURCE RESERVATION

The Resource Reservation Tool provides the interface for reserving IP resources for devices or applications in an oversubscribed system. After IP resources are reserved for particular purposes, the remaining resources are shared on a first-come, first-served basis. See **page 647** for details.

IP DEVICE RESOURCE MANAGER

The IP Device Resource Manager maintains all IP resource requests in the system along with all allocated IP resources in the system. See **page 1119** for details.

DATA CONNECTIONS

The following data connections are maintained by the Private IP Networking feature:

- **Data Channels**: The system maintains one data connection between each node programmed for Private IP Networking. The system uses this data connection for call control and datagram messaging. When a data connection fails, the system attempts to establish a new connection using alternative IPR application resources if they are available. If all data connections fail, the system may drop calls in progress, and further call attempts using the connection are blocked.

- **Route Optimization**: Call control route optimization using IP works almost identically to call control route optimization using T1/E1/PRI. The system identifies instances where a call passes through the same node twice (a route loop) and eliminates the loop. The system releases all resources within the loop, including DSPs.

AUDIO CONNECTIONS

The system monitors the following audio connections:

- Audio Connection Status
- Audio Connection Parameters
- Audio Route Optimization
- Insufficient Bandwidth

See **page 1116** for details.

IP CONNECTIONS AND IP CONNECTION GROUPS

The following are IP Connection and Node IP Connection Group descriptions.

- **IP Connections**: Contain the programming information for each IP connection in the network. The extensions associated with each connection are automatically added to this folder when local or remote IP connections are created. Programmable information for each connection includes IP addresses, gateway addresses, subnet masks, backplane configurations, and so forth. For programming instructions, see "System and Device IP Settings" on **page 599**.

- **Node IP Connection Groups**: Correspond to an IP network connection between the remote node and the local node. Node IP Connection Groups are automatically added for each remote node created. Programmable information for each Node IP Connection Group includes IP call configurations, emergency outgoing access lists, outgoing access lists, and so forth. For programming instructions, see "Node IP Connection Group" on **page 627**.

QOS

The system sets the ToS (Type of Service) bits in the IP packet headers to aid the network in implementing Quality of Service (QoS). The system uses the Audio RTP Type of Service and Data Type of Service flags to set the appropriate ToS bits in the header. For programming instructions, see "Local Processor Module and Expansion Card IP Settings" on **page 617**.

AUDIO CONNECTIONS TO IP DEVICES

The system recognizes that both IP phones and voice resources on IPR application resources configured for private networking require different audio connection parameters than other devices in the system.

The affected connection parameters include:

- **Gain Values**: For the purpose of gain adjustment, the system recognizes IP phones and IP connections as IP devices and adjusts the gain settings accordingly.

- **Enhanced Speakerphone Mod**e: When a phone connects to an IP phone or to a voice resource on the IPR application, the system does not allow the phone to enter enhanced speakerphone mode. Enhanced speakerphone mode does not function properly when the audio travels over an IP network.

NOTE	Sometimes the system may not detect that a call is connected over an IP network. If this situation occurs, the system may allow users to enable enhanced speaker phone mode, which will degrade the call quality.

DTMF CONFIGURATION

You can program the IPR application to use a specific set of values that affect DTMF detection. This configuration allows you to make the system more or less sensitive to DTMF tones, which ultimately affects the amount of talkoff. In general, the more sensitive the system is to DTMF tones, the more likely talkoff is to occur. For programming instructions, see "DTMF Detection Information" on **page 934**.

DIAGNOSTICS

The IP Networking application includes several tools for performing diagnostics when troubleshooting. These tools are separate from call processing and can display only information local to the IPR application. There are also programming displays to control how the call behaves.

The four groups of diagnostics tools are as follows:

- **Web/HTML**: Contains displays that have a graphical interface for ease of use.

- **Message Print**: Allows you to see any information, warning, or debug messages when a predetermined condition, such as a bus error, occurs.

- **Exception Log**: Is a collection of Exception Log entries that signify major violations, errors, or events. Major resets, minor resets, code-checker exceptions, bus errors, and unhandled exceptions are examples of Exception Log entries.

- **IPR Application Logs**: Executed from the olm_bin directory, application diagnostic commands provide IPRA diagnostic information in log format.

NAT TRAVERSAL FOR IP PHONES

Network Address Translation (NAT) is a method for translating internal IP addresses or network addresses into public Internet addresses.

The MiVoice Office 250 supports connecting with IP phones over the Internet while allowing networking to utilize a customer's wide area network (WAN) behind a Network Address Translation (NAT)/firewall. This system enhancement was not available in the initial v1.0 release.

This enhancement permits a system residing behind a NAT/firewall to utilize a static IP address to allow IP phones from the Internet to traverse, or bypass, the NAT. This capability is useful because NAT does not translate IP addresses in the payloads of IP packets, and VoIP traffic on the Internet is subject to Quality of Service (QoS) problems.

The MiVoice Office 250 can be programmed to use the appropriate address for each IP phone— Native (internal to the WAN) or NAT (external to the WAN). In other words, instead of depending on NAT, the MiVoice Office 250 can be programmed to use the appropriate IP address when communicating with external IP phones.

For instructions to implement near-NAT traversal for IP phones, see "NAT Address Type" on **page 244**.

The preferred method to deploy remote phones (teleworker phones) is to connect a Mitel Border Gateway to the MiVoice Office 250.

For more information about networking options in general, refer to "Appendix B: Network IP Topology," in the *MiVoice Office 250 Installation Manual* .

AUTOMATIC NAT DETECTION

NOTE	Automatic NAT Detection applies to older legacy phones only, and is not supported on the new 53xx and 86xx serise IP phones.

Automatic Network Address Translation (NAT) Detection allows Mitel IP phones to operate inside or outside a private network NAT or firewall without having to change the **NAT Address Type** field in the IP Settings folder in DB Programming every time the phone is relocated. The feature enables Mitel hard IP phones and the Mitel 8602 IP softphone to place and receive calls either from inside or outside a Mitel private network. An example NAT configuration is shown in the figure on page 115.

NAT TRAVERSAL FOR IP PHONES

The MiVoice Office 250 supports the connection of IP phones from the public network when the MiVoice Office 250 is located inside a local area network (LAN).

To accomplish this near-end NAT traversal capability, the MiVoice Office 250 is programmed with both its Native IP address and the corresponding statically NATed public IP address. The platform being programmed with both addresses allows system-defined IP phones calling in from the Internet to traverse the NAT/firewall.

AUTOMATIC DETECTION FOR NAT TRAVERSAL

The Automatic NAT Detection feature takes near-end NAT traversal one step further. An IP phone designated with the **Auto** NAT Type in DB Programming automatically selects the appropriate NAT or Native IP address. This feature allows the IP phone to be moved from inside a NAT to outside and back again without programming intervention.

If the default **Auto** NAT Type is designated for an IP phone in DB Programming, the Auto NAT Detection feature determines the correct IP address to use when the phone comes online. Once the NAT type is determined, the IP phone uses the setting for all calls. However, the NAT IP address associated with the system must be programmed manually in order for the NAT detection test to run.

This feature does *not* work with:

- SIP Trunks
- MGCP Gateways
- MGCP phones
- 52xx\53xx phones
- IP Gateways
- IP trunks

- Mitel wireless 8664, 8665, and 8668 phones

- Mitel IP phones in SIP mode

- non-Mitel SIP phones.

Firmware on these phones cannot run the Auto NAT Detection test. For these phones, manually select the NAT Type setting in DB Programming.

Neither automatic nor manually selected NAT traversal applies to the 8601 SoftPhone for Pocket PC application.

The figure below shows the configurations of phones programmed for the three NAT Types in DB Programming.
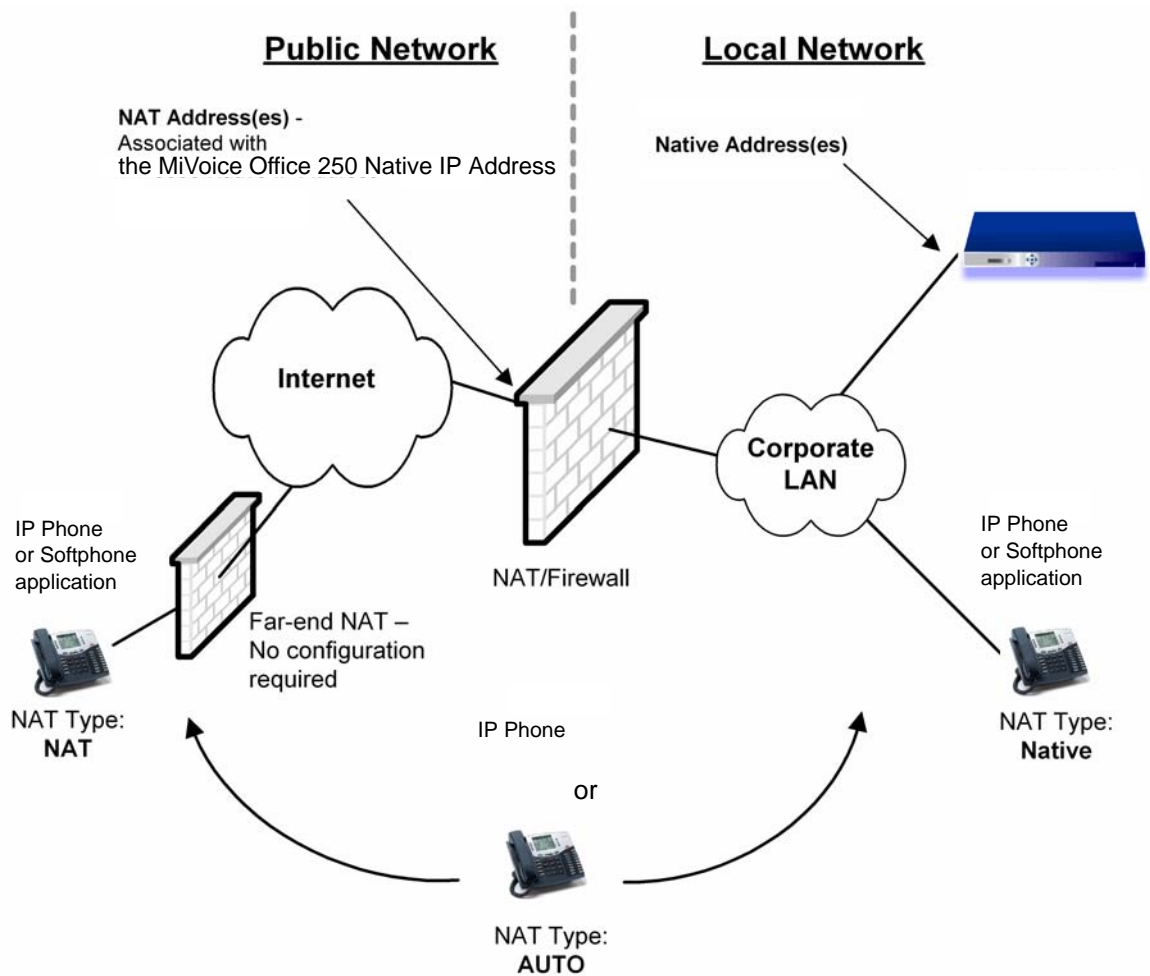
COMPONENT COMPATIBILITY FOR 99 NODES SUPPORT

This section contains the following information:

- **99 Nodes Support Component Compatibility Table** below

- Troubleshooting the 99 Nodes Features on **page 117**

NOTE	Due to licensing changes in v4.0, an Extended Node Capacity license is no longer required to support above 63 nodes.

99 NODES SUPPORT COMPONENT COMPATIBILITY TABLE

The table below lists component version numbers that have been validated to support the 99 Nodes system feature.

NOTE	Mitel recommends that you use these versions if you are using the 99 Nodes feature, so you should upgrade to the latest non-chargeable upgrade available.

MIVOICE OFFICE 250 VERSION	V2.3	V2.4	V3.0	V3.1	V3.2	V4.0 OR LATER
Applications						
CT Gateway	4.30x				4.40x or Later	
Database Programming	2.3.6.46	2.4.x.x	3.0.x.x	3.1.x.x	3.2.x.x or Later	
System Manager	2.0.6.102	2.0.6.104 or Later				
Auxiliary Applications						
Attendant Console	3.004	3.1x or Later				
Contact Center Suite	4.11.0030		4.2.1.x or Later			
Customer Service Manager	(see Contact Center Suite)			5.0.0.x or Later		
Network Qualifier	2.006 or Later					
Unified Communicator®	4.0.x.x or Later					
Unified Communicator for the MiVoice Office 250	(see Unified Communicator)			5.0.x.x or Later		
Unified Communicator Advanced	Not Supported				3.0.x.x or Later	
Unified Messaging Open Standards Edition (OSE)	1.0	2.0 or Later				
Unified Messaging v2.2 (VPU)	2.222	Not Supported				
Unified Messaging v2.3	2.365b	Not Supported				
Devices						

MIVOICE OFFICE 250 VERSION	V2.3	V2.4	V3.0	V3.1	V3.2	V4.0 OR LATER
52XX/53XX	Not Supported				1.06.00.04 or Later	
8528/8568	Not Supported				All Versions	
8660	8.2.2		8.3.0 or Later			
8664/8665/8668	1.0.0.4 or Later					
8600/8620/8662	2.2.02		2.2.04 or Later			
8690	1.1.8 or Later					
IP PhonePlus	8.1.0 or Later					
IP SLA	8.1.0		8.2.0 or Later			

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TROUBLESHOOTING THE 99 NODES FEATURES

Table 18 contains troubleshooting information for 99 Nodes support.

Table 18:   99 Nodes Support Troubleshooting Tips

Symptom	Possible Cause	Corrective Action
Increased Node Capacity: A system is continuously experiencing software exceptions and resetting. The system resides in a network of 63+ nodes. The node is running earlier than v2.1 software.	The node that is resetting is running an invalid software version. After the network is operating at a 63+ node network, nodes cannot downgrade to earlier than v2.1.	Upgrade the node to v2.1 or later.
Attendant Console Causes Slow-Downs: The system slows down when an Attendant Console application connects to a system.	This is a "brute force" OAI command that requests all the off-node device information from the system. The slow down is directly related to the number of off-node devices in the system.	Leave the Attendant Console running continuously rather than shutting off the application during non-work hours. This will reduce the number of refreshes the console performs to get status of the devices in the network.
"Brute Force" Network Broadcasts: The system uses the last IP resource. The system sends a broadcast message to all nodes in the network to inform them that it cannot handle IP networking calls. As an IP resource frees up, the system sends a broadcast message to all nodes in the network again to inform them that it can handle IP networking calls. If the system oscillates between this last IP resource, it can cause a tremendous amount of IP traffic on the entire network.	Insufficient IP networking resources.	This condition of oscillating on the last IP resource is undesirable. If the system is in this state, upgrade to a Processor Expansion card to increase the number of IP resources, or attempt to reduce the number of IP phones to allow more networking IP resources. Note that v2.1 or later system has been modified to prevent the "brute force" network broadcasts.

TROUBLESHOOTING IP NETWORKING ISSUES

This section contains the programming tips for the following issues:

- **IP Networking** below

- Network Node on **page 121**

IP NETWORKING

Table 19 summarizes the troubleshooting strategies recommended for resolving discrepancies that may occur with IP networking.

Table 19: IP Networking Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Cannot make calls across the IP network	No VoIP channels are available.	Make sure you have the proper networking resources allocated.
	The remote IP address is programmed incorrectly.	Make sure the off-node IP connections have the correct Remote IP Address programmed. See "Remote Node IP Connections" on **page 626**. This address must match the IP address programmed locally on that node.
	The node cannot find the IP connection.	Make sure the correct Node IP Connection Group is added to the appropriate node. See "Remote Node IP Connections" on **page 626**.
No audio on network calls	Improper configuration of IP connections, firewall and/or network address translation (NAT)	Refer to Appendices A and B for network configuration guidelines in the *MiVoice Office 250 Installation Manual* .
	The audio stream receive ports are not programmed correctly.	Make sure the audio stream ports fields match for the affected nodes. See "Local Processor Module and Expansion Card IP Settings" on **page 617**.
Phone display shows INSUFF BANDWIDTH FOR VOICE	The data network does not meet the minimum specifications required for IP networking.	Run the Network Qualifier to assess your data network. For more information about the Network Qualifier, refer to the *Network Qualifier Reference Manual*.
	The data network is experiencing problems (for example, excessive packet loss and traffic).	Contact your network administrator.
One-way audio on network calls	Improper configuration of IP connections, firewall and/or network address translation	See Appendices A and B in the *MiVoice Office 250 Installation Manual* for network configuration guidelines.
	The audio stream receive ports are not programmed correctly.	Make sure the audio stream ports fields match for the affected nodes. See "Local Processor Module and Expansion Card IP Settings" on **page 617**.

Table 19:   IP Networking Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Poor call quality on network calls	The data network is experiencing problems (for example, excessive packet loss and traffic).	Contact your network administrator.
	The data network does not meet the minimum specifications required for VoIP.	Run the Network Qualifier to assess your data network. Then verify that your data network meets the minimum specifications. For more information about the Network Qualifier, refer to the *Network Qualifier Reference Manual*.
	The G.711 vocoder is using too much bandwidth.	Switch to a G.729 vocoder, which uses less bandwidth.
	The G.729 vocoder is compressing audio data, resulting in poorer quality.	Switch to a G.711 vocoder, which does not compress audio data, resulting in higher call quality.
	Choppy audio	Adjust the Audio Frames per Packet in DB programming. Try adjusting it up or down to see if it improves audio quality.
DTMF tones are not detected	DTMF tone length is too short.	Set the DTMF Digit Duration/Pause field to 100ms or higher. See "Timers and Limits" on **page 756**.
	The wrong DTMF vocoder is programmed.	Make sure the DTMF Encoding field is set to either G.711 or RFC2833. See "DTMF Encoding Setting" on **page 635**.
Cannot fax across the network	IP faxing resources are not allocated appropriately.	Make sure you do not have more IP fax resources allocated than networking resources.
	Insufficient bandwidth	Run Network Qualifier to determine the bandwidth. For more information about the Network Qualifier, refer to the *Network Qualifier Reference Manual*.
The network is experiencing audio and/or connection problems	There is a port conflict or the firewall, NAT, or router is blocking the port.	Try changing the ports associated with IP call control and/or audio. Make sure none of them conflict with ports that other protocols use (for example, SIP uses 5060). See Appendices A and B in the *MiVoice Office 250 Installation Manual* for network configuration guidelines.
	Improper configuration of IP connections, firewall and/or network address translation.	See Appendices A and B for network configuration guidelines.
	Insufficient bandwidth	Run Network Qualifier to determine the bandwidth. For more information about the Network Qualifier, refer to the *Network Qualifier Reference Manual*.

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Table 19:   IP Networking Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Echo is heard (half-duplex)	Noise levels are too high.	Talk at a lower volume while on a call.
	Callers are talking at the same time (doubletalk).	When simultaneous and similar audio on both sides of the call is detected, the application may try to cancel this perceived "echo," causing audio and echo problems. Try to avoid doubletalk.
	The data network does not meet the minimum specifications required for IP networking.	Run the Network Qualifier to assess your data network. Then verify that your data network meets the minimum specifications. For more information about the Network Qualifier, refer to the *Network Qualifier Reference Manual*.
	The backplane configuration settings (echo settings in particular) are not optimized.	Change the backplane configuration settings. See the "Echo Profiles" on **page 735**.
Cannot call devices on other nodes after configuring IP Network using the Networking Wizard.	Wrong Off-Node IP Connection IP Address.	Check IP Connection IP Addresses on the local and the remote nodes.
	Missing IP Connection Group or Off-Node IP Connection.	Verify that there's a Node IP Connection Group in the target node's Node Trunk/IP Connection Groups folder. Make sure there's an Off-Node IP Connection in that NIPCG, and verify the IP Address.
	Missing IP Networking Port License(s) or Private Networking software license.	Check the License and software license features for Networking.
	WAN connection is down.	Check your WAN connection.
Cannot call devices on other nodes after configuring T1/ E/PRI Networking using the Networking Wizard.	No B-Channels.	Make sure the T1M or T1M-2 module is licensed for PRI, is online, and the appropriate B-Channels are equipped.
	No Reference Clock.	Check reference clock list on this and other nodes. Make sure one node is master for private and the other is slave to private.
Cannot import/export to/ from new target node after using the Networking Wizard.	Networking is not programmed correctly.	Verify that there's a Node Trunk/IP Connection Group to the remote node.
		Verify that the Node Trunk/IP Connection Group includes the Node Trunk Group or Off-Node IP Connection to the remote node. Depending on the type of programming.
		Refer to the Help file for each programming area.

Table 19:   IP Networking Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
No audio on a call between two IP devices (in the same Network Group) configured to use peer-to-peer media.	The Network Group is set for peer-to-peer media, but there is a firewall or NAT in between phones.	Verify that there is no firewall or NAT between the two phones in the Network Group. See Appendices A and B in the *MiVoice Office 250 Installation Manual* for network configuration guidelines.
	Improper configuration of IP connections, firewall and/or network address translation	See Appendices A and B for network configuration guidelines.
	A Network Group is programmed to use peer-to-peer media on one node, but not on another node.	Verify that the Use Peer-to-Peer Media flag is set to **Yes** on all nodes.
	A Network Group is not configured to use peer-to-peer media.	Verify that the Use Peer-to-Peer Media flag is set to **Yes** on all nodes.

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NETWORK NODE

Table 20 summarizes the troubleshooting strategies recommended for resolving discrepancies that may occur with a network node.

Table 20:   Network Node Troubleshooting Chart

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Cannot access a phone on another node. Display shows OUTGOING ACCESS DENIED	The phone or application making the call does not have outgoing access for the Node Trunk Group that it needs	Program a Node Trunk Group, if one does not exist. Program outgoing access for the phone or application. You can use an extension list to make this task easier. See "Extension Lists" on **page 500**.
Cannot access a phone on another node. Display shows DESTINATION UNREACHABLE	There is no IP connection.	For IP networking, make sure you can ping both nodes. For PRI networking, make sure the connection is there and the span is up.
	There is no Node IP Connection Group programmed for the node.	Program a "P6XXX" Node IP Connection Group for the networked nodes. See "Node IP Connection Group" on **page 627**.
	A single port is busied out on a node.	For example, a system has networked node 1 and node 2. When a single port is busied out on node 1, node 2 will not be notified that the channel has been busied out. That same port must be manually busied out on node 2. For more information about specifying Busy Out commands for ports, see "Busy Out Manager" on **page 1098**.

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Table 20:   Network Node Troubleshooting Chart (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
When calling a phone on another node, using the Automated Attendant, calls go directly to voice mail without trying the extension number first	The off-node device is not associated with a mailbox of the same number	Make sure the mailbox is associated with a corresponding extension number. If so, make sure the off-node device exists on the node attached to the external voice processing unit using the Export/Import feature. Then delete the off-node device for the phone that was associated with the mailbox and create a new off-node device with a temporary extension. Change the extension number of the temporary off-node device to the extension number associated with the mailbox.
When using a PRI span on another node, calls do not go out properly (for example, callers hear ringing but there is no answer)	The PRI span does not have the "CO Provides Tones" flag enabled	Enable the CO Provides Tones flag for the span. See "System Flags" on **page 751**.
Cannot establish an IP connection	NATs/firewalls are present.	See Appendices A and B in the *MiVoice Office 250 Installation Manual* for network configuration guidelines.
	The IP connection extension for the node is not unique.	Make sure the IP connection extensions (P8XXY) are unique.
	There is no Node IP Connection Group programmed for the node.	Program a "P8XXX" Node IP Connection Group for the networked nodes. See "Node IP Connection Group" on **page 627**.
	The IP address and subnet mask information for each IPRA is not valid.	Make sure each IPRA has the correct IP address and subnet mask programmed. See the "System IP Settings" on **page 603**.

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Chapter 5

NUMBERING PLANS

INTRODUCTION

This chapter provides information to program MiVoice Office 250 Numbering Plan options, as shown in the figure below. The system Numbering Plan determines how the system manages outgoing calls. Numbering Plan programming options include the following:

- Area Flags on **page 125**

- Classes of Service (COS) on **page 127**

- Device Baseline Extensions on **page 132**

- Automatic Route Selection (ARS) on **page 134**

  - Dial Rules

  - Facility Groups

  - Route Groups

- Emergency Calls on **page 150**

- Home Area Codes on **page 152**

- Toll Strings on **page 153**

- User Groups on **page 156**
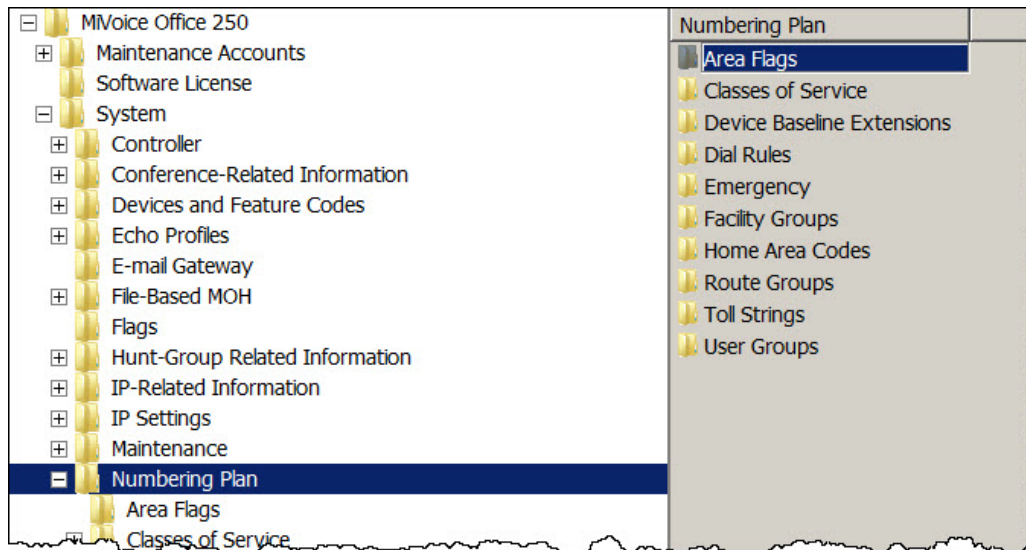
AREA FLAGS

U.S. installations only. If your system office codes (digits 4, 5, and 5 in a 10-digit number) and area codes (digits 1, 2, and 3 in a 10-digit number) overlap, or if toll digits are allowed on toll calls or long distance calls, you can change the area and office code flags.

The first two flags from the bulleted list below determine how area and office codes overlap. **Table 21** shows the difference between the standard North American Numbering Plan (NANP) and each of the overlap flags, which are represented by the following variables:

- N = 2–9

- Z = 0 or 1

- X = 0–9

Table 21:   NANP and Overlap Flag Differences

NANP/OVERLAP FLAG	AREA CODES CAN BE:	OFFICE CODES CAN BE:
NANP	NZX	NXX
Office Codes as Area Codes	NZX or NXX	NXX
Area Codes as Office Codes	NZX	NXX or NZX

The following are system Area Flags used for Numbering Plans:

- **Office Codes Used as Area Codes**: An area code in another location uses an NXX pattern that matches an office code within the system site area code. Because the system cannot differentiate between an office code and an area code when the second digit dialed is 0–9, it will wait for the Interdigit timer to expire or another digit to be dialed before assuming that dialing is completed.

- **Area Codes Used as Office Codes**: One or more office codes within the system site area code use an NZX pattern that is the same as an area code in another area. Because area codes do not resemble office codes (NXX), end-of-dialing detection is not affected by this flag.

- **Local 7/10 Digit Dialing**: When this flag is enabled, outgoing calls are identified as having reached the end of dialing if the first digits are not a toll field, equal access field, operator access field, or a local area code. This function speeds up placement of local seven-digit calls in an area where some local calls require 10 digits.

- **Toll Digit Allowed On Toll Local Calls**: This option applies only if the area and office codes overlap. Callers in the site area code usually dial a 1 when placing a call within the local area code(s).

- **Toll Digit Required On Toll Long Distance Calls**: This option applies only if the area and office codes overlap. Callers in the site area code must dial a 1 when placing a call outside of the local area code(s).

To enable or disable a Numbering Plan flag:

1. Select System – Numbering Plan – Area Flags – *<**flag**>*.

2. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

3. Click out of the field or press **ENTER** to save the change.

CLASSES OF SERVICE (COS)

This section contains the following information:

- **Feature Description** below

- Programming COS Options on **page 129**

FEATURE DESCRIPTION

Class of Service (COS) is used for toll restriction, which prevents system users from placing outgoing calls. COS designations (01–09 or 01–07) have default values and COS designations. COS designations 10–16 or 08–16 are blank. Classes of service 02–16 have programmable dialing patterns, and all have programmable day and night lists of devices.

Exact (complete) pattern matches with classes of service marked as "allowed" always override exact (complete) pattern matches with classes of service marked as "denied." Also, partial (incomplete) pattern matches with classes of service marked as "allowed" always override partial (incomplete) pattern matches with classes of service marked as "denied." The only time a pattern in a denied class of service overrides a pattern in an allowed class of service is when the match with the denied pattern is exact (complete) and the match with the allowed pattern is

COS FOR U.S. SYSTEMS

The following are COS designations for U.S. systems (for European systems, see **page 128**):

- **COS 01 – ARS Only:** *(This is a phone COS only. It is not used for trunk groups.)* When enabled, users must use Automatic Route Selection (ARS) to place calls. Users hear re-order tones when attempting to place a call using any other method. A restricted user can still select individual trunks if the trunks are designated as "exempt from ARS Only" (see **page 512**), were transferred, were placed on hold, or are recalling or ringing. Trunk restriction determines which trunks in the ARS route group can be selected by the phone or voice processing application.

- **COS 02 – Deny Area/Office:** This restriction is divided into eight user groups (see **page 156**) to allow the use of varying area and office code restriction tables. This reduces restrictions for some of the phones, voice processing applications, or trunk groups while increasing restrictions for others. Each phone, application, and trunk group is assigned a day mode and a night mode user group. Within each user group, you can designate area codes as restricted, allowed, or extended. Restricting an area code prevents users from placing calls to that area code. Allowing an area code allows all office codes within that area code. You can designate an area code as extended to determine which office codes (up to 800) are allowed or restricted within that area code. For each user group, you can mark up to 800 area codes as allowed or restricted in the database list, and up to six area codes can be marked as extended.

- **COS 03 – Deny Operator:** Calls to numbers that match the dial patterns for this class of service (defaults to [Q]RN+, [Q]R0, and [Q]RE) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the phone, voice processing application, or trunk group.

- **COS 04 – Deny Toll Access:** Calls to numbers that match the dial patterns for this class of service (defaults to [Q]TN+ and [Q]TE) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the phone, voice processing application, or trunk group.

- **COS 05 – Deny International:** Calls to numbers that match the dial patterns for this class of service (defaults to [Q]I+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the phone, voice processing application, or trunk group.

- **COS 06 – Deny Equal Access:** Calls to numbers that match the dial patterns for this class of service (defaults to Q+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the phone, voice processing application, or trunk group.

- **COS 07 – Deny Local Calls:** Calls to numbers that match the dial patterns for this class of service (defaults to N+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the phone, voice processing application, or trunk group.

- **COS 08 – Denied Numbers:** Calls to numbers that match the dial patterns for this class of service (defaults to 1900NXXXXXX+ and 976XXXX+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the phone, voice processing application, or trunk group being used. Calls are only restricted if the dialed patterns match the denied pattern exactly and that is the only class of service you have. Allowed numbers (as follows) always override denied patterns, even if the numbers are similar.

- **COS 09 – Allowed Numbers:** Calls to numbers that match with the dial patterns for this class of service, defaults to 1(800, 888, 877, 866, 855, 844, 833, and 822)NXXXXXX+, are allowed, even if number also matches a dial pattern in a restricted class of service that is assigned to the phone, voice processing application, or trunk group being used.

COS FOR EUROPEAN AND AUSTRALIAN SYSTEMS

The following are COS designations for European and Australian systems (for U.S. systems, see **page 127**):

- **COS 01 – ARS Only:** *(This is a phone COS only. It is not used for trunk groups.)* Calls can only be placed using the Automatic Route Selection (ARS) feature when this restriction is assigned. The user will hear reorder tones when attempting to place a call using any other method. A restricted user can still select individual trunks if the trunks are designated as "exempt from ARS Only" (see **"Toll Restrictions"** on **page 511**), were transferred, were placed on hold, or are recalling or ringing. Trunk restriction determines which trunks in the ARS route group can be selected by the phone or Voice Processing application.

- **COS 02 – Deny Operator:** Calls to numbers that match the dial patterns for this class of service (defaults to R+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the phone, voice processing application, or trunk group being used.

- **COS 03 – Deny Toll Access:** Calls to numbers that match the dial patterns for this class of service (defaults to TN+, TE, 010+, and T1+) are restricted, unless the number also

matches a dial pattern in an "allowed" class of service that is assigned to the phone, voice processing application, or trunk group being used.

- **COS 04 – Deny International:** Calls to numbers that match the dial patterns for this class of service (defaults to I+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the phone, voice processing application, or trunk group being used.

- **COS 05 – Deny Local Calls:** Calls to numbers that match the dial patterns for this class of service (defaults to N+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the phone, voice processing application, or trunk group being used.

- **COS 06 – Denied Numbers:** Calls to numbers that match the dial patterns for this class of service (defaults to 0891+ and 0898+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the phone, voice processing application, or trunk group being used. Calls are only restricted if the dialed patterns match the denied pattern exactly and that is the only class of service you have. Allowed numbers (as follows) always override denied patterns, even if the numbers are similar.

- **COS 07 – Allowed Numbers:** Calls to numbers that match with the dial patterns for this class of service, defaults to 0345+, 0500+, 0645+, and 0800+, are allowed, even if number also matches a dial pattern in a restricted class of service that is assigned to the phone, voice processing application, or trunk group being used.

PROGRAMMING COS OPTIONS

Program the COS options as described in the following sections.

CHANGING OR ADDING COS DESCRIPTIONS

You can change or add COS descriptions.

To change or add a COS description:

1. Select System – Numbering Plan – Classes of Service– <***class of service***>.

2. Select the current description, and then type the new description in the box.

3. Click out of the field or press **ENTER** to save the change.

ADDING DEVICES TO COS DAY AND NIGHT LISTS

To add a device to the COS Day or Night list of devices:

1. Select System – Numbering Plan – **Classes of Service**.

2. Double-click the COS designation.

3. Double-click either **Day** or **Night**.

4. Right-click anywhere in the right pane, and then click **Add To Day** (or **Night**) **List**. A window appears prompting for the device type to include.

5. Select the device types (you can use the SHIFT or CTRL key to select more than one item), and then click **Next**. The items with details appear. To view items in a list only, click **List**.

6. Select the appropriate items, and then select **Add Items**. When you have added all the devices, click **Finish**. The selections appear in the list. To view programming options, double-click the extension number.

To deleted a device from the COS Day or Night list of devices:

Select the device, right-click, and then select **Remove Selected Items**.

DEFINING COS DIALING PATTERNS

(COS 03–16 only). You can define COS dialing patterns.

To add a dialing pattern to the class of service:

1. Select System – Numbering Plan – **Classes of Service**.

2. Double-click the COS designation.

3. Double-click **Dialing Patterns** to view the current list.

4. Do one of the following:

To add a dialing pattern to the bottom of the list:

a. Do not select any existing patterns.

b. Right-click anywhere in the right pane, and then select **Add To List**. A blank pattern appears at the bottom of the list.

To add to the list above an existing device:

a. Select the device below the location where you want the new entry.

b. Right-click and select **Add To List**. A blank pattern appears above the pattern you selected.

To delete a COS dial pattern:

Select the dialing pattern, right-click, and then select **Remove Selected Items**. (You can use the SHIFT or CTRL key to select more than one item.)

To move a pattern to another location in the list:

Do one of the following:

- Drag and drop the dial pattern to the new position

- Select the dial pattern to move and press **CTRL** + the up/down arrow to move the dial pattern up or down in the list.

PROGRAMMING DIALING PATTERN DIGITS

To program dialing pattern digits:

1. Select System – Numbering Plan – **Classes of Service**.

2. Double-click the COS designation.

3. Double-click **Dialing Patterns** to view the current list.

4. In the **Dialing Pattern** column, type the new digits in the box.

5. Click out of the field or press **ENTER** to save the change.

ALLOWING OR RESTRICTING DIALING PATTERNS

If you want the dialing patterns to be "allowed," enable this option. If you want the dialing patterns to be "restricted," disable the option.

To allow or restrict a dialing pattern:

1. Select System – Numbering Plan – **Classes of Service**.

2. Double-click the COS designation.

3. Select **Allow Dial Patterns**.

4. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

5. Click out of the field or press **ENTER** to save the change.

DEVICE BASELINE EXTENSIONS

Device Baseline Extensions define the recommended starting extension numbers for new system devices. See **Figure** .

NOTE	To support the MiCollab Unified Messaging (UM) voice processing system, Device Baseline Extensions include new Session Initiation Protocol (SIP) options. For more information about MiCollab Unified Messaging (UM), see **page 836**.

When you add a system device, the system checks the Device Baseline Extension list for the starting extension, as shown in the following example.
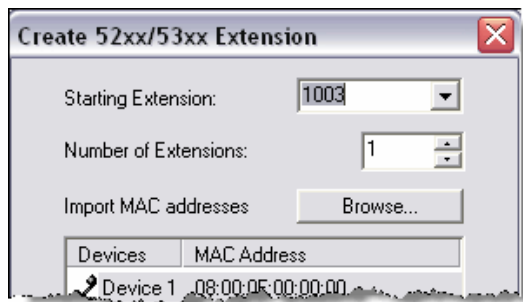
In this example, the Device Baseline Extension for a phone is 1000. However, extensions 1000 and 1001 were previously assigned, so the system shows the first available extension, extension 1003, in the Starting Extension box. The number of digits is maintained. That is, if the Device Baseline Extension were 10000 instead, the system would show the first available extension after 10000. Note that wildcard extensions (for example, 12XXX) are not valid.

To change a Device Baseline Extension:

1. Select System – Numbering Plan – **Device Baseline Extensions**. A list of the system Device Baseline Extensions appear in the right pane.

2. In the **Extension** column, select the current extension, and then type the new extension number in the box.

3. Click out of the field or press **ENTER** to save the change.

You can also use the Configuration Wizard to program the following devices with starting baseline extensions:

- Phones
- Hunt Groups
- Voice Processor Applications
- Page Zones
- Trunk Groups
- Trunks

See the *MiVoice Office 250 Database Programming Help* for details.

AUTOMATIC ROUTE SELECTION (ARS)

This section contains the following information:

- **Feature Description** below

- Programming ARS on **page 138**

FEATURE DESCRIPTION

Automatic Route Selection (ARS) is a money-saving feature that allows the system to be programmed to select the least expensive route for placing outgoing calls. It can be used for placing outgoing calls and transferring or forwarding calls to outside telephone numbers. Phones can be restricted to using only ARS for placing outgoing calls. Also, because users do not have direct access to trunks on other nodes, ARS is the only way users can place calls using the other nodes' trunks.

NOTE	This feature requires a software license.

Each node has its own Automatic Route Selection (ARS) programming, which the system uses to select the least expensive route for outgoing calls. Because users do not have direct access to trunks on other nodes, ARS is the only way users can place calls to trunks on other nodes.

ARS calls are limited to one "hop" to another node. For example, if the system routes an outgoing call to another node, the other node cannot route the outgoing call to any other node. This prevents the possibility of an infinite loop when the system searches for a node to route the outgoing call.

Make sure the phones that use ARS have outgoing access for the trunk groups and nodes. For more information about trunk group programming, see "Programming Trunk Group Options" on **page 506**. For more information about node trunk group programming, see "Viewing or Changing Node Trunk Group Information" on **page 548**.

You program ARS using route groups and facility groups with dialing rules:

Route Groups: A route group contains dialing patterns and facility groups.

- The dialing patterns are used to determine the calls that will be routed through the route group. For example, the default dial pattern for Route Group 1 is N+, any number of digits beginning with digit 2–9 for the U.S. (digit 2–9, 345+, 0500+, 0645+, and 0800+ for Europe). If a number is dialed that begins with 1, it will not be routed through this route group.

- Each route group has an ordered list of facility groups that contain lists of local trunk groups and/or nodes. There can be 100 facility groups in the system. You should program facility groups so that the least-expensive route is checked and, if available, is selected first. If the least-expensive facility group is not available the system checks the other groups in the list until it finds an available trunk.

Facility Groups: A facility group contains trunk group or node lists and dial rules.

- The list can include local trunk groups or nodes. They cannot contain node trunk groups.

- The dial rules tell the system what to dial. The system can have up to 32 dial rules, 26 of which are programmable. Each facility group can use up to 32 dial rules. For example, if

the selected route group requires that the number contain "1" but no area code (national dialing in Europe), the dial rules include the 1 and drop the area code (national dialing). The modified number can contain up to 32 digits. (If SMDR is enabled, the modified number, not the digits dialed, will appear in the SMDR call record.) When programming ARS, you can use preset dial rules or create new dial rules that add up to 16 digits each. For more information about Dial Rules, see "Programming ARS Dial Rules" on **page 139**.

When ARS is selected, the user dials the number—including the Area Code, if needed—and the system performs the following actions:

1. **Checks the dialed number and matches the dialing pattern to a route group**: The system checks the route groups in numerical order and selects the first group that applies to the dialing pattern of the number that was dialed.

> **NOTE** If the Emergency Call feature is programmed to use ARS, the emergency calls will always go through Route Group 1, regardless of route group programming.

2. **Selects a facility group**: If all of the trunks are busy, and the phone is enabled for ARS camp on, the call will camp on to the facility group until a trunk is available.

3. **Checks for toll restrictions and outgoing access**: Before the number is modified by the facility group's dial rules, the system checks the phone toll restrictions and outgoing access to determine whether the call is allowed. (All calls placed using ARS are toll restricted, regardless of whether the selected trunk is subject to toll restriction. However, Emergency Call feature calls are never toll restricted.) If allowed, the system continues to the next step. If not allowed, the system sends reorder tones and the call is not placed.

4. **Adds or deletes digits according to the facility group chosen**: Each facility group has a programmed set of dial rules that tells the system what to dial. For example, if the selected route group requires that the number contain "1" but no area code, the dial rules include the 1 and drop the area code.

5. **Dials the modified telephone number**: If the number is allowed, the system seizes an idle trunk in one of the selected trunk groups, waits for the Dialing Wait After Connect timer to expire, and then dials the number.

When ARS is used, the phone user hears dial tone when the feature code is entered (manually or automatically using a CALL button, the ARS button, or the **OUTGOING** button). The user then dials the number and hears silence until the trunk is seized and dialing is completed. (The user will not hear the digits being dialed.) The call appears under an available CALL button or trunk button.

AUTOMATIC ROUTE SELECTION EXAMPLE

Phone user enters 1234 account code and dials 1-602-961-9000

SYSTEM

CO

OUTGOING CALL
DIALED =
1-602-961-9000-1234

② ARS does the following:

- Checks the number and finds dialing pattern 602+ in Route Group 2.
- Selects a facility group with an available trunk.
- Checks toll restriction and outgoing access.
- Uses dial rules to modify the number:
  Echo 3 Digits After Toll
  Echo Local Address
  Add Account Code
- Dials the modified number.

ARS ON A NETWORK

Each node has ARS programming. All trunk groups and nodes in a facility group must reside on the same node as the facility group. Using ARS is the only way a user can access trunks on other nodes. If the system routes an outgoing call to another node using an intermediary node, the intermediary node ARS cannot use the nodes in its facility group to move the outgoing call to another node. This eliminates the system from getting into an infinite loop searching for a node to route the outgoing call.

When determining toll restriction for an ARS outgoing call, the network checks the phone toll restriction based on the database information on the node on which the phone resides, not the node that contains the trunk which the system uses to place the call. The system does not check the trunk COS for ARS calls.

For the purposes of the telephone call cost display and SMDR output, the network computes call cost using the factors and rates on the node on which the outgoing trunk resides. In other words, if a phone on Node A dials what is considered a long-distance number on Node A, and the call is routed to Node B where the number is considered local, the phone will see the local call cost rate which equals the value on Node B.

SAMPLE ARS CALL ON A NETWORK

① Phone user dials 1-602-961-9000

NODE 1 (IN 714 AREA CODE)

NODE ROUTE GROUP TO NODE 2
Digits sent **1-602-961-9000**

NODE 2 (IN 602 AREA CODE)

LOCAL TRUNK CO

④ Local call placed to 961-9000

② ON NODE 1: ARS routes call to Node 2 echoing all of the digits (no other dial rule modification). Call is sent using the route group with dial pattern 602+ and the facility group with route to Node 2.

③ ON NODE 2: ARS routes call to its Local route group (dial pattern 602+) and uses dial rules to remove the toll field and area code (use only the Echo Local Address dial rule). Call is placed using an available trunk in the Local route group's facility group.

ATTENDANT RECALL

When a call is placed on hold or is transferred from one phone to another, the Hold timer and the Transfer timer limit the amount of time the call may remain unattended. After that timer expires, the call recalls the phone that transferred it or placed it on hold, and the Recall timer starts, which leads to one of the following possible events:

- If the call remains unanswered at the phone until the Recall timer expires, it recalls the phone's attendant and the Abandoned Call timer starts. If the system provides no attendant, the call continues to recall at the phone that transferred it or placed it on hold.

- If the attendant phone is busy, the call camps on, and the display shows the source of the recall.

- If the call is not answered before the Abandoned Call timer expires, the system disconnects the call.

If an attendant phone transfers a call that is not answered before the Attendant Transfer timer expires, it will recall the attendant phone.

If a phone user transfers or forwards an outside call to an outside telephone number, it becomes an "unsupervised" CO-to-CO call because no inside parties are involved. The CO-to-CO call is limited by the Unsupervised CO timer. When the timer expires, the call recalls the primary attendant phone and causes the **CNF** button to flash. (Display phones show UNSUPERVISED CNF RECALL.) This serves two purposes:

- It allows the attendant to monitor the length of CO-to-CO calls. When a CO-to-CO call recalls, the attendant can disconnect the call or allow it to continue.

- If the callers hang up before the attendant receives the recall, the system may not have disconnected the trunks because a disconnect was not received from the CO. The attendant must disconnect the call.

When a recall rings at the attendant's phone, a CALL or individual trunk button flashes at a medium rate.

If the attendant has calls forwarded, recalls from phones follow internal call forward requests. Recalls do not forward to outside telephone numbers but recall the attendant's phone until they are answered or the Abandoned Call timer expires. Placing the attendant's phone in Do-Not-Disturb mode does not block recalls or direct ring-in calls.

If there is not a primary attendant, recalls remain at the phones and ring until answered or disconnected by the system.

To answer a hold or transfer recall at an attendant phone:

When you see a hold or transfer recall display (XXXX RCL FROM YYYY) and hear a recall ringing, lift the handset. Or, do one of the following:

- **Outside call recall**: Press the medium-flashing CALL button or individual trunk button. Or, press the **ANSWER** button. (If more than one trunk is recalling, pressing the **ANSWER** button selects the outside call indicated on the display.)

- **Intercom call recall**: Press the **IC** button or the **ANSWER** menu button. (If you are busy when the intercom call recalls, it will camp on. The **IC** button flashes at the medium rate, but you do not hear recall ring signals.)

- **Conference call recall**: Press the flashing **CNF** button to connect with the conference call. The **CNF** button flashes slowly, and the display shows CONFERENCE IN PROGRESS. *If the parties are still talking,* press the **CNF** button again and hang up to return the parties to their conversation. The **CNF** button flutters. You can enter the conference at any time by pressing the fluttering **CNF** button. If the Hold timer expires, the conference recalls your phone again. *If the parties have hung up,* hang up to disconnect the call.

PROGRAMMING ARS

The following sections detail ARS programming.

- **Planning ARS Requirements** below
- Programming ARS Dial Rules on **page 139**
- Programming ARS Facility Groups on **page 141**
- Programming ARS Route Groups on **page 145**

PLANNING ARS REQUIREMENTS

The first step of implementing ARS for your system is to determine your requirements.

To determine your ARS requirements:

1. Determine the types of calls that system users will make (for example, local, long distance, international). Where will the call destinations be? You need a route group and a facility group for each type of call.

2. Determine the dialing patterns for each type of call. For example, local calls begin with any digit 2–9 (but *not* a toll digit) and operator assisted calls begin with 0 (1XX in Europe). These are the dialing patterns that tell the system which route group to select.

3. Determine the best routes for each type of call based on where the system is installed.

   Examples:

   - *U.S. systems:* What are the best routes for each type of call? If you have nodes in Phoenix and Los Angeles, would it be better to route calls from Phoenix to Southern

California through a node or over your long distance service? This will determine the trunk groups and nodes included in each facility group.

- *European systems:* What are the best routes for each type of call? If you have nodes in London and Kettering, would it be better to route calls from London to Glasgow through a node or over your long distance service? This will determine the trunk groups and nodes included in each facility group.

4. Determine the facility group order. That is, determine the trunks on this node or other nodes that would be the best route for each type of call.

5. Determine facility groups dial rules. That is, determine what special characters, if any, need to be added or removed from the dialed number when calls are placed? Determine this for each trunk you are using.

Examples:

- *U.S. systems*: If a caller in Phoenix dials 1-714-XXX-XXXX, the system must remove the toll digit and possibly the area code if the call is sent out over local trunks on the Los Angeles node. However, if the call is routed through the long distance provider in Phoenix, you probably need to include (echo) the toll and area code digits or even add other digits as required by the long distance service.

- *European systems*: If a caller in London dials 020-8335XXX, the system must remove the toll digit and possibly the national dialing code if the call is sent out over local trunks on the Kettering node. However, if the call is routed through the long distance provider in London, you will probably need to include (echo) the toll and national dialing code digits or even add other digits as required by the long distance service.

PROGRAMMING ARS DIAL RULES

After you determine the ARS requirements for your organization (see **page 138**), program the dial rules. Each facility group (see **page 141**) includes dial rules that determines system dialing. For example, if the selected facility group requires that the number contain the digit "1" but no area code (national dialing in Europe), the dial rules include (echo) the toll field, but would not echo the area code. When programming ARS, you can use preset dial rules or create new dial rules. The system can have up to 32 dial rules, 26 of which are programmable. The following are the default dial rules:

U.S. Systems:

- **Dial Rule 1 – ECHO Equal Access**: Includes the equal access digits (wildcard Q, which defaults to 10XXX and 101XXX) in the number, if dialed.

- **Dial Rule 2 – ECHO Toll Field**: Includes the toll field (1, 0, 01, or 011) in the number, if dialed.

- **Dial Rule 3 – ECHO 3 Digits After Toll Field**: Includes the three digits after the toll field in the number. These digits are usually the area code.

- **Dial Rule 4 – ECHO Local Address**: Allows ARS to dial the rest of the digits that were dialed by the phone user. Non-programmable.

- **Dial Rule 5 – ECHO Account Code**: Causes the system to dial the account code that is associated with the call, if available before end of dialing. The account code can be entered using any of the account code types, including All Calls Following, as long as the system

receives the account code before end of dialing. To use an optional account code, the code must be entered before the number is dialed. Non-programmable.

- **Dial Rule 6 – ECHO Extension Number**: Requires the system to include the extension number of the phone being used to place the call. Non-programmable.

- **Dial Rule 7 – ADD #**: Adds a pound/hash (#) to the dialed number. This dial rule is programmable.

European Systems:

- **Dial Rule 1 – ECHO Toll Field**: Includes the toll field in the number, if dialed.

- **Dial Rule 2 – ECHO Local Address**: Allows ARS to dial the rest of the digits that were dialed by the phone user.

- **Dial Rule 3 – ECHO Extension Number**: Tells the system to include the last three digits of the extension number of the phone being used to place the call.

- **Dial Rule 4 – ECHO Account Code**: Causes the system to dial the last three digits of account code that is associated with the call, if available, before end of dialing. The account code can be entered using any of the account code types, including All Calls Following, as long as the system receives the account code before end of dialing.

- **Dial Rule 5 – ECHO Serial Number**: Tells the system to dial the system-assigned serial number of an ARS call. The serial number range is 000–998, excluding 112. The serial number can be reset in dial rule programming.

- **Dial Rule 6 – ADD #**: Adds a hash (#) to the dialed number. This dial rule is programmable. Dial Rules 3 (Echo Extension), 4 (Echo Account Code), and 5 (Echo Serial Number) will *always* come after Dial Rules 1 and 2. If 3, 4, or 5 precedes Dial Rules 1 or 2, the SMDR Call Type and Call Cost will be affected.

Australian Systems:

- **Dial Rule 1 – Echo Toll**: Includes the toll field in the number, if dialed.

- **Dial Rule 2 – ECHO Local Address**: Allows ARS to dial the rest of the digits that were dialed by the phone user.

- **Dial Rule 3 – ECHO Extension Number**: Tells the system to include the last three digits of the extension number of the phone being used to place the call.

- **Dial Rule 4 – ECHO Account Code**: Causes the system to dial the last three digits of account code that is associated with the call, if available, before end of dialing. The account code can be entered using any of the account code types, including All Calls Following, as long as the system receives the account code before end of dialing.

- **Dial Rule 5 – ECHO Serial Number**: Tells the system to dial the system-assigned serial number of an ARS call. The serial number range is 000–998, excluding 112. The serial number can be reset in dial rule programming.

- **Dial Rule 6 – ADD #**: Adds a hash (#) to the dialed number. This dial rule is programmable. Dial Rules 3 (Echo Extension), 4 (Echo Account Code), and 5 (Echo Serial Number) will *always* come after Dial Rules 1 and 2. If 3, 4, or 5 precedes Dial Rules 1 or 2, the SMDR Call Type and Call Cost will be affected.

To program ARS dial rules:

NOTE	You cannot edit the dial rule description (Dial Rule field).

1. Select System – Numbering Plan – **Dial Rules**. The list of rules 1–32 is shown in the right pane. You can program rules 7–32, allowing you to add digits to a dialed number.

2. In the **Digits** column, type the digits for the dial rule. Dial rules can contain any digit (0–9, *, #) hookflashes, and pauses. The number can have up to 32 digits.

3. *If you are programming dial rules for a U.S. system,* go to **step 5**.

4. *If you are programming dial rules for a European system,* complete the following:

    a. *(For European systems only)*. In the **Hidden** column, enable or disable the Hidden flag to determine if the dial rules will be hidden. It is set to No by default.

    b. *(For European systems only)*. In the **Absorbed** column, enable or disable the Absorbed flag to determine if the dial rules will be hidden. It is set to Yes by default.

    c. *(For European systems only)*. In the **ARS Serial Number** column, enter the serial number (000-998, excluding 112) that you are using. This is only programmable for Dial Rule 5 – ECHO Serial Number.

5. Click out of the field or press **ENTER** or to save the change.

PROGRAMMING ARS FACILITY GROUPS

Facility groups determine the trunk groups that will route calls and the required dial rules. All facility groups are programmable. The default facility group values are as follows.

U.S. systems:

*   **Local (P1500)**: Uses Trunk Group 1 and Dial Rules 3 and 4.

*   **Toll Local (P1501)**: Uses Trunk Group 1 and Dial Rules 2 and 4.

*   **Toll Long Distance (P1502)**: Uses Trunk Group 1 and Dial Rules 1, 2, 3, and 4.

*   **Operator (P1503):** Uses Trunk Group 1 and Dial Rules 1, 2, 3, and 4.

*   **International Station-to-Station (P1504**): Uses Trunk Group 1 and Dial Rules 1, 2, 3, 4, and 7.

*   **International Operator (P1505**): Uses Trunk Group 1 and Dial Rules 1, 2, 3, 4, and 7.

European systems:

*   **Local (P1500)**: Uses Trunk Group 1 and Dial Rules 1 and 2.

*   **National (P1510)**: Uses Trunk Group 1 and Dial Rules 1 and 2.

*   **Operator (P1512)**: Uses Trunk Group 1 and Dial Rules 1 and 2.

*   **International (P1513)**: Uses Trunk Group 1 and Dial Rules 1 and 2.

Australian Systems:

*   **Local (P1500)**: Uses Trunk Group 1 and Dial Rules 1 and 2.

- **Toll Long Distance**: Uses Trunk Group 1 and Dial Rules 1 and 2.

- **Operator (P1512)**: Uses Trunk Group 1 and Dial Rules 1 and 2.

- **International (P1513)**: Uses Trunk Group 1 and Dial Rules 1 and 2.

- **Unassigned/Default**: Uses Trunk Group 1 and Dial Rules 1 and 2.

Each facility group has a list of trunk groups and/or node trunk groups for routing calls. For example, the "Local" facility group would contain trunk groups that include local trunks, but a "Los Angeles" facility group might have node trunks groups that include local trunks on a node in Los Angeles.

You must place facility groups in the in the order that you want them to be selected. Make sure that the facility groups contain the proper type of trunks (local, FX, WATS, long distance, and so forth) for the calls that are routed.

Make sure you do the following for each facility group:

- Determine the following facility group options based on your ARS requirements (see **page 138**).

- Determine what types of calls will be processed and program names for those groups.

- Determine which trunk groups and nodes will be used and program the list. Make sure that the trunk groups and nodes contain the proper type of trunks (local, FX, WATS, long distance, and so forth) for the calls that will be routed (use the information from steps **3** and **4** on **page 139**. Also, make sure the phones that will be using ARS have outgoing access for the trunk groups and nodes. For more information, see **"CO Trunk Groups"** on **page 504** and "Node Trunk Groups" on **page 548**.

- Program the list of dialing rules so that the outgoing numbers include all of the necessary digits. See **page 143**.

CREATING AND DELETING FACILITY GROUPS

To create a facility group:

1. Select System – Numbering Plan – **Facility Groups**.

2. Right-click in any blank area in the right pane.

3. Select **Create Facility Group**.

4. Enter the starting extension number and number of extensions, and then click **OK**. The new facility group appears in the list.

5. If applicable, select the description, and then type the new description in the box.

6. Click out of the field or press **ENTER** to save the change.

To delete a facility group:

Select the facility group, right-click, and then select **Delete**.

PROGRAMMING ARS FACILITY GROUP DIALING RULES

You must assign dial rules to each facility group.

To program ARS facility group options:

1. Select System – **Numbering Plan**.

2. Double-click **Facility Groups**. The current list of facility groups appears in the right pane.

3. Double-click **Dial Rules**. The current list of dial rules, if any, appears. This is an *ordered list*. Place the dial rules in the list in the order you want them to be used.

4. Do one of the following

    To add to the bottom of the list:

    a. Do not select any existing dial rules.

    b. Right-click anywhere in the right side pane. An option box appears.

    To add to the list above an existing dial rule:

    *   Select the dial rule below the location where you want the new entry, and then right-click. An option box appears.

5. Select **Add To Dial Rules List**. A window appears prompting for the types to include.

6. Select the type (you can use the SHIFT or CTRL key to select more than one item), and then click **Next**. The items with details appear. To view items in a list only, click **List**.

7. Select the appropriate items, and then select **Add Items**. When you have added all dial rules, click **Finish**. The selections appear in the list. To view programming options, double-click the dial rule.

To move a dial rule to another location in the list:

Do one of the following:

*   Drag and drop the dial rule to the new position

*   Select the dial rule to move and press **CTRL** + the up/down arrow to move the dial rule up or down in the list.

To delete one or more dial rule from the list:

Select the dial rule, right-click, and then select **Remove Selected Items**.

PROGRAMMING TRUNK GROUPS AND NODES FOR FACILITY GROUPS

Trunk groups and nodes must be placed in the list in the order that they should be selected. Make sure that the trunk groups or nodes being used in the facility group contain the proper type of trunks (local, FX, WATS, long distance, and so forth) for the calls that will be routed. Trunk group toll restrictions will not apply when ARS is used; only phone toll restrictions are checked.

To add the trunk groups and/or nodes to be used by this facility group:

1. Select System – **Numbering Plan – Facility Groups**. The current list of facility groups appears in the right pane.

2. Double-click the facility group.

3. Double-click **Trunk Groups/Nodes**. The current list of trunks/nodes, if any, appears. This is an ordered list. Place the trunks and nodes in the list in the order you want them to be used.

4. Do one of the following

   To add to the bottom of the list:

   a. Do not select any existing dial rules.

   b. Right-click anywhere in the right side pane. An option box appears.

   To add to the list above an existing dial rule:

   • Select the trunk group or node below the location where you want the new entry, and then right-click. An option box appears.

5. Select **Add To Trunk Groups/Nodes List**. A window appears prompting for the device types to include.

6. Select **CO Trunk Group** and/or **Remote Node**, and then click **Next**. The list of available devices with details appears. To view devices in a list only, click **List**.

7. Select the appropriate items, and then select **Add Items**. When you have added all dial rules, click **Finish**. The selections appear in the list. To view programming options, double-click the dial rule.

To delete dial rules from the list:

Select the dial rule, right-click, and then select **Remove Selected Items**.

To move a trunk group or node to another location in the list:

Do one of the following:

• Drag and drop the trunk group or node to the new position

• Select the trunk group or node to move and press **CTRL** + the up/down arrow to move the trunk group or node up or down in the list.

To delete a trunk group or node:

Select the trunk group or node, right-click, and then select **Remove Selected Items**.

PROGRAMMING ARS ROUTE GROUPS

Make sure you do the following for each ARS route group:

- Determine what types of calls will be processed by each route group and program names for those groups. See step 1 on **page 138**.

- Program dial patterns (see **page 139**). Dial patterns determine which route group to select, as described in step 2 on **page 138**. When checking the route group, the system looks at the first dial pattern. If it does not match, the system continues to check the other patterns in the route group list. If a match is found, the route group is selected. If not, the next route group is checked.

- Select the facility groups to route the calls. See **page 148**.

DEFAULT ROUTE GROUPS

All route groups are programmable. The default values are as follows:

U.S. systems:

- **Local (P1000)**: Used for all calls that do *not* begin with a toll digit (1) or operator digit (0) and for calls to Emergency Numbers (its default dial pattern is N+). It uses the Local facility group.

> **NOTICE**
>
> When ARS is used to place an emergency call (see **page 150**), the system uses Route Group 1, even if it contains nodes. This means that the network *can* access a trunk on a node other than the user's node if the user accesses ARS and dials the emergency number. Mitel highly recommends that local trunks be installed and used for emergency number trunk access and that nodes are *not* used in Route Group 1*.*

- **Toll Local (P1003)**: Used for calls with a toll digit (1) and a seven-digit number, and for calls to 1N11 (N=Any digit 2–9). Its default dial patterns are TN11 and TXXXXXXX. It uses the Local Facility Group.

- **Toll Long Distance (P1011)**: Processes calls with a toll digit (1) and a 10-digit number. The numbers may also include equal access digits. The default dial pattern is [Q]TNXXXXXX+. This Route Group uses the Toll Long Distance Facility Group.

- **Operator (P1013)**: Processes calls that begin with an operator digit (0) but do *not* begin with an international access code (01 or 011). The numbers may also include equal access digits. Its default dial patterns are [Q]RN+, [Q]RR, and [Q]R. This Route Group uses the Operator Facility Group.

- **International Station-to-Station (P1014)**: Used for calls that begin with an international phone-to-phone access code (011) but do *not* begin with the international operator access code (01). The numbers may also include equal access digits. The default dial pattern is [Q]011+. This Route Group uses the International Station Facility Group.

- **International Operator (P1015)**: Used for calls that begin with an international operator access code (01). The numbers may also include equal access digits. The default dial pattern is [Q]01+. This Route Group uses the International Operator Facility Group.

European and Australian systems:

- **Local (P1000)**: Used for all calls that do *not* begin with a toll digit (01 or 0) or operator digit (1XX) and for calls to emergency services.

> **NOTICE**
>
> When ARS is used to place an emergency call (see **page 150**), the system uses Route Group 1, even if it contains nodes. This means that the network *can* access a trunk on a node other than the user's node if the user accesses ARS and dials the emergency number. Mitel highly recommends that local trunks be installed and used for emergency number trunk access and that nodes are *not* used in Route Group 1.

- **Toll [National] (P1011)**: Processes calls with a toll digit (0 or 01). This Route Group uses the National Facility Group.

- **Operator (P1013)**: Processes calls that begin with an operator digit (1XX) but do *not* begin with an international access code (00).

- **International (P1014)**: Used for calls that begin with an international access code (00). Dial patterns are assigned to the route group in the order they will be used. When checking the route group, the system will look at the first dial pattern. If it does not match, the system will continue checking the other patterns in the route group list. If a match is found, the route group is selected. If not, the next route group is checked.

CREATING OR DELETING ARS ROUTE GROUPS

This procedure can be done only if there are fewer than 32 route groups.

To create a route group:

1. Select System – **Numbering Plan**.
2. Double-click **Route Groups**. The current list of facility groups appears in the right pane.
3. Right-click in any open area in the right pane.
4. Select **Create Route Group**. The new route group appears at the bottom of the list.
5. If applicable, change the description of the route group by selecting its current description, and then typing the new information in the box.
6. Press **ENTER** or select another field to save the change.

To delete a route group:

Select the group to be removed, right-click, and then click **Delete**.

MOVING ROUTE GROUPS IN LISTS

To move a route group to another location in the list:

Do one of the following:

- Drag and drop the route group to the new position.
- Select the route group to move and press **CTRL** + the up/down arrow to move the route group up or down in the list.

PROGRAMMING ARS ROUTE GROUP DIAL PATTERNS

The dial pattern tells the system which route group to select. When checking the route group, the system looks at the first dial pattern. If it does not match, the system continues to check the other patterns in the route group list. If a match is found, the route group is selected. If not, the next route group is checked.Only the default route groups have default dial patterns, as summarized in **Table 22** and **Table 23**. A complete list of special characters and toll strings is shown on **page 153**.

Table 22:   Route Group Dial Patterns for U.S. Systems

ROUTE GROUP #	DIAL PATTERN(S)
Local (1001)	N+
Toll Local (1004)	TN11, TXXXXXXX
Toll Long Distance (1011)	[Q]TNXXXXXX+
Operator (1013)	[Q]RN+, [Q]RR, [Q]R
International Station-to-Station (1014)	[Q]011+
International Operator (1015)	[Q]01+

Table 23:   Route Group Dial Patterns for European Systems

ROUTE GROUP #	DIAL PATTERN(S)
Local (1001)	N+
Toll Local (1004)	TN11, TXXXXXXX
Toll Long Distance (1011)	[Q]TNXXXXXX+
Operator (1013)	[Q]RN+, [Q]RR, [Q]R
International Station-to-Station (1014)	[Q]011+
International Operator (1015)	[Q]01+

To program ARS route group dial pattern:

1. Select System – **Numbering Plan** – **Route Groups**. The current list of route groups is displayed.

2. Double-click **Dial Patterns**. The current list of dial patterns, if any, appears.

3. Right-click anywhere in the right pane, and then click **Add To Dial Patterns List**. The Batch Create Patterns dialog box appears.

4. Type the digits or special characters that apply to all patterns created in this screen.

5. Select the number of patterns to create.

6. Click **OK**. The new dial patterns are added to the list, each reflecting the specified base pattern. For example, if you entered 1480 as the base pattern and you selected 5 as the number of patterns, five patterns, each with 1480, is added to the list.

7. Press **ENTER** or select another field to save the change.

To move a dial pattern to another location in the list:

Do one of the following:

- Drag and drop the dial pattern to the new position.

- Select the dial pattern to move and press **CTRL** + the up/down arrow to move the dial pattern up or down in the list.

To delete one or more dial pattern from the list:

Select the dial pattern, right-click, and then select **Remove Selected Items**.

ADDING FACILITY GROUPS TO ARS ROUTE GROUPS

For each route group, select the facility group(s) to be used to route the calls. Facility groups must be placed in the list in the order that you want them to be selected.

To add facility groups that will be used by this route group:

1. Select System – **Numbering Plan** – **Route Groups**. The current list of route groups is displayed.

2. Double-click **Facility Groups**. The current list of facility groups, if any, appears. This is an *ordered list*. Place the facility groups in the list in the order you want them to be used.

3. Do one of the following

   To add to the bottom of the list:

   a. Do not select any existing dial rules.

   b. Right-click anywhere in the right side pane. An option box appears.

   To add to the list above an existing dial rule:

   - Select the facility group below the location where you want the new entry, and then right-click. An option box appears.

4. Select **Add To Facility Groups List**. A window appears prompting for the device type to include.

5. Select **Facility Group**, and then click **Next**. A list of available facility groups with details appears. To view facility groups in a list only, click **List**.

6. Select the facility groups, and then click **Add Items**. If you select more than one, they are placed in the list in numerical order, *not* in the order you select them. To keep the list in the proper order, you may have to add one at a time.

7. When you have added all dial rules, click **Finish**. The selections appear in the list. To view programming options, double-click the dial rule.

To move a facility group to another location in the list:

Do one of the following:

- Drag and drop the facility group to the new position.

- Select the facility group to move and press **CTRL** + the up/down arrow to move the facility group up or down in the list.

To delete a facility group from the list:

- Select the facility group, right-click, and then click **Remove Selected Items**.

PROGRAMMING AUDIO FOR CALLS CAMPED ONTO THIS DEVICE FOR ARS ROUTE GROUPS

Audio for Calls Camped onto this Device defines the audio that a caller hears when camped-on to the route group. For more information about audio settings, see **page 414**.

To program the Audio for Calls Camped onto this Device option:

1. Select System – Numbering Plan – Route Groups.

2. Double-click the route group.

3. Select **Audio for Calls Camped onto this Device**.

4. In the Value column, select the option from the list.

5. Click out of the field or press **ENTER** to save the change.

EMERGENCY CALLS

The Numbering Plan Emergency option is where you program the emergency numbers that the system uses when users enter the Emergency Call feature code (for example, 911 in the U.S.).

WARNING

POSSIBLE DELAY IN LOCAL EMERGENCY RESPONSE TO REMOTE SITES.

IP and SIP phone users should be alerted to the following hazardous situations:

- If an Emergency Call phone number is dialed from an IP or Session Initiation Protocol (SIP) phone located at a remote site that is not equipped with a correctly configured gateway, the call will be placed from the location where system chassis is installed rather than from the location where the emergency call is made.

  In this situation, emergency responders may be dispatched to the wrong location. To minimize the risk of remote site users misdirecting emergency responders, Mitel recommends regular testing of MGCP/SIP gateway trunks for dial tone.

- If uninterruptible power supply (UPS) protection has not been installed as part of the MiVoice Office 250, IP and SIP phones will *not* operate when electrical power fails either at remote sites or at the main system location.

To place calls during a power failure in this situation, IP and SIP phone users can only use a single line phone connected to one of the power failure bypass circuits built into the system chassis. If a phone connected to a power failure bypass circuit is not available, users should make emergency calls **from a local phone *not* connected to the system**. For details about the Power Failure Bypass feature, refer to the Installation chapter in the *MiVoice Office 250 Installation Manual* .

NOTICE

It is the responsibility of the organization and persons performing the installation and maintenance of Mitel Advanced Communications Platforms to know and comply with all regulations required for ensuring Emergency Outgoing Access at the location of both the main system and any remote communication phones. Remote IP and SIP phones may require gateway access to nearby emergency responders. Emergency Call phone numbers include:

- 911, the default for Mitel systems located in the U.S.
- 999, the default for Mitel systems located in the European market and used primarily in the U.K.
- If applicable, 112, an emergency number used widely in Europe outside of the U.K, and Australia.

Any emergency number, such as for a police or fire station, that is appropriate for the location of the main system and/or remote phones.

Emergency calls, by default, use the first local trunk group and are not sent through other nodes using node trunk groups. However, when ARS is used to place an emergency call, Route Group 1 is used even if it contains nodes. This means that the network *can* access a trunk on a node other than the user's node if the user accesses ARS and dials the emergency number. *Mitel highly recommends that local trunks be installed and used for emergency number trunk access and that nodes are not used in Route Group 1.* When a user places an emergency call, every administrator in the network receives an emergency alarm.

You can store up to 10 emergency numbers. For more information about the emergency call feature, see "Emergency Calls" on **page 182**.

To program or change emergency numbers:

1. Select System – Numbering Plan – **Emergency**.

2. Double-click **Emergency Numbers**.

3. Select the current value or blank field of the number you want to program.

4. Type the number, up to 48 digits, in the box. Only the first emergency number has a default value (911 in the U.S., 999 in the UK, or 112 in Europe outside of the UK, and Australia).

5. Click out of the field or press **ENTER** to save the change.

HOME AREA CODES

U.S. installations only. The database requires a "home" area code for identifying local calls. It also allows up to 15 additional (codes 2–15) home area codes. The system refers to these area codes for toll restriction and call cost.

NOTE	The first area code is an area code that is always stripped from the dialing number. Other home area codes serve to identify local calls, but are still dialed. If local 10-digit dialing is always required despite the area codes, do not list an area code in the first home area code. If a return call (callback feature) from voice mail is not working correctly, refer to the Knowledge Base article 1224 for possible solutions. See "Technical Support" on **page 20**.

To program the home area code(s) for the system site:

1. Select System – Numbering Plan – **Home Area Codes**.

2. Select the home area code option (1–16).

3. In the **Digits** column, type the new area code in the box.

4. Click out of the field or press **ENTER** to save the change.

TOLL STRINGS

This section contains the following information:

- **Feature Description** below

- Programming Toll Strings on **page 154**

FEATURE DESCRIPTION

Toll strings are dialing patterns that are abbreviated to single character "wildcards." Wildcards are used in COS and ARS programming. Toll strings can contain any digit from 0 through 9 and the keypad special characters # and *. You can also use a variety of characters to represent particular digit strings, hookflashes (recalls in Europe), or special digit strings. Each of the following toll string wildcards can be reprogrammed or renamed for your system.

NOTE	Changing toll string dialing patterns affects all of the other parts of the system where they are used.

Table 24 shows the programmable toll string wildcards for U.S. systems.

Table 24:   Toll String Wildcards for U.S. systems

CHARACTER	MEANING
S	**Any Toll String**: Any of the designated patterns can be dialed at that point in the number. In the default state, the dialing patterns for this toll string are "I," "R," and "T."
R	**Operator Access**: Represents the digit string that is required to call the service provider's operator. In the default state, this dialing pattern is 0.
T	**Toll Access**: Represents the digit(s) required for using long distance service. In the default state, this dialing string is 1.
I	**International**: Represents the digits required for international dialing. In the default state, the dialing patterns for this toll string are 011 and 01.
Q	**Equal Access**: Represents the digits required to select secondary carriers for equal access dialing. In the default state, the dialing patterns for this toll string are 101XXXX, 10NXX, and 100XX.

Table 25 shows the programmable toll string wildcards for European systems.

Table 25:   Toll String Wildcards for European Systems

CHARACTER	MEANING
S	**Any Toll String**: Any of the designated patterns can be dialed at that point in the number. (In the default state, the dialing patterns for this toll string are "I," "R," and "T.")
R	**Operator Access**: Represents the digit string that is required to call the network provider's operator. (In the default state, this dialing pattern is 1XX.)
T	**Toll Access:** Represents the digit(s) required for using long distance service. (In the default state, this dialing strings are 0 and 01.)
I	**International**: Represents the digits required for international dialing. (In the default state, the dialing pattern for this toll string is 00.)

Table 26 shows the special characters that may be used when specifying dialing patterns. These characters are *not* programmable.

Table 26:   Special Characters for Dialing Patterns

CHARACTER	MEANING
X	Any digit 0–9
A	Any keypad entry (0–9, #, *)
N	Any digit 2–9
Z	Any digit 0 or 1
B	# or *
H	A hookflash (recall)
E	End of dialing; the pattern will not match if any other digits are dialed beyond this point
+	Any additional dialing will be accepted, from that point in the string, with no further checking for a match. This also means that no further dialing is required beyond this point.
[*x*]	Indicates an optional pattern within another pattern. For example, with a U.S. system, the International Access character (I) could be defined as 01[1]. (The 01 is followed by an optional 1.)
(*x-x*)	Indicates a range of digit strings within a pattern. The strings on either side of the hyphen and all strings that fall within the numerical range are included in the match. The strings on either side of the hyphen must be the same length, and the only digits that may appear in the range are 0–9 (#, *, pauses, and flashes are not allowed).
<*x*>	Indicates repeatable patterns within patterns. In other words, no matter how many times the digit string within the brackets is dialed, the system considers the dialed digits to match the pattern. Note that a repeatable pattern is an entire pattern; no other characters are allowed before or after a repeatable pattern. In other words, a repeatable pattern cannot be included within any other pattern.

PROGRAMMING TOLL STRINGS

To program toll strings:

1. Select System – Numbering Plan – **Toll Strings**. Toll strings are shown in the right pane.

2. If desired, select the current description, and then enter the new description in the box.

3. Click out of the field or press **ENTER** to save the change.

To move a dial pattern to another location in the list:

Do one of the following:

• Drag and drop the dial pattern to the new position.

• Select the dial pattern to move and press **CTRL** + the up/down arrow to move the dial pattern up or down in the list.

To program the digits for a pattern:

1. Select the current value for the pattern, and then enter the new digits in the box.

2. Click out of the field or press **ENTER** to save the change.

ADDING OR DELETING TOLL STRING DIAL PATTERNS

To add or delete a toll string dial pattern:

1. Select System – Numbering Plan – **Toll Strings**. Toll strings are shown in the right pane. This is an *ordered list.* Place the devices in the list in the order you want them to be accessed.

2. Do one of the following

To add to the bottom of the list:

    a. Do not select any existing dial patterns.

    b. Right-click anywhere in the right pane, and then click **Add to List**.

3. *To add to the list above an dialing pattern*:

Select the device below the location where you want the new entry, right-click, and then click **Add To List**. A blank pattern appears above the pattern you selected.

To delete one or more patterns from the list:

1. Double-click the toll string to view the current pattern(s).

2. Select the item(s), right-click, and then select **Remove Selected Items**. (You can press the SHIFT or CTRL keys to select more than one item.)

USER GROUPS

This section contains the following information:

- **Feature Description** below

- Programming Area Codes on **page 156**

FEATURE DESCRIPTION

U.S. installations only. This section describes how to program area and office code restriction used for the Deny Area/Office class of service.

You can set up area and office code tables of up to eight user groups to allow different area/office code restriction to be used. This is useful for reducing restrictions for some users while increasing restrictions for others. Each phone, application, and trunk group is assigned to a user group.

Planning User Groups

Within each user group, area codes can be restricted, allowed, or extended:

- Restricting an area code prevents users from placing calls to that area code and all of its office codes.

- Allowing an area code allows all office codes within that area code.

- Designating an area code as "extended" allows you to determine which office codes within that area code are allowed or restricted. Up to six extended area codes can be identified within each user group.

To prepare for programming:

1. List the area codes that are allowed, restricted, and extended. Also, list the office codes within the extended area codes that are allowed and restricted.

2. Select System – Numbering Plan – **User Groups** to view a list of the eight user groups. The list shows the user group numbers and their descriptions, if programmed. At this level, you can program the description. To program the area codes, office codes, day list, and night list for a user group, double-click the user group that you want to program.

3. Select the current description, and then enter a name for the user group, up to 20 characters, in the text box.

4. Click out of the field or press **ENTER** to save the change.

PROGRAMMING AREA CODES

You can drag and drop area codes from one location to another. This ensures that each area code is include in a list and none are skipped. To select a series of items, hold down **SHIFT** while selecting the first and last items in the range. To select two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

This section contains the following information:

- **Programming Allowed Area Codes** below
- **Programming Extended Area Codes** below
- Programming Restricted Area Codes on **page 158**
- Programming the Area Code Day/Night List on **page 158**

PROGRAMMING ALLOWED AREA CODES

To allow users in the user group access to area codes, the codes must be placed in the Allowed list.

To place an area code in the Allowed list:

1. Select System – Numbering Plan – **User Groups**.
2. Double-click the user group.
3. Double-click **Extended** or **Restricted Area Codes**, and then locate the area code that you want to restrict.
4. Drag and drop that area code to the Allowed area codes directory in the left pane. The next time you open Allowed area codes, the codes will be in the directory.

To remove an area code from the Allowed list:

Move the area code to the Restricted or Extended list in the user group.

PROGRAMMING EXTENDED AREA CODES

You can use up to six extended area codes for each user group, and each of the six can support an individual list of allowed and restricted office codes.

To create an Extended Area Code:

1. Select System – Numbering Plan – **User Groups**.
2. Double-click the user group.
3. Double-click **Allowed** or **Restricted Area Codes**, then locate the area code you want to extend.
4. Drag and drop the area code to the Extended Area Codes directory in the left pane. The next time you open Extended Area Codes, the codes will be in the directory. Office codes can then be programmed as allowed or restricted for that area code by dragging and dropping them into the appropriate categories.

To remove an area code from the Extended list:

Move the area code to the Restricted or Allowed list in this user group.

PROGRAMMING RESTRICTED AREA CODES

You can prevent users from placing calls to certain area codes.

To place area codes in the Restricted list:

1. Select System – Numbering Plan – **User Groups**.

2. Double-click the user group.

3. Double-click **Allowed** or **Extended Area Codes**, and then locate the user group that you want to restrict.

4. Drag and drop the area code to the Restricted Area Codes directory in the left portion of the window. The next time you open Restricted Area codes, the codes will be in the directory. To remove an area code from the Restricted list, move it to the Allowed or Extended list in this user group.

PROGRAMMING THE AREA CODE DAY/NIGHT LIST

Only extensions with COS 02 can be placed in a user group day or night list.

To program the area code Day or Night list:

Method A

Drag and drop the phone to the appropriate list. You can only move from Day list to Day list or Night list to Night list. You cannot move phones between Day and Night lists.

Method B

1. Select System – Numbering Plan – **User Groups**.

2. Double-click the user group.

3. Right-click anywhere in the right pane, and then click **Move To List**. A window appears prompting for the device type to include.

4. Select the device types (you can use the SHIFT or CTRL key to select more than one item), and then click **Next**. The items with details appear. To view items in a list only, click **List**.

5. Select the appropriate items, and then select **Move Items**. When you have added all the devices, click **Finish**. The selections appear in the list. To view programming options, double-click the extension number.

To delete an item from the list:

You must move the item to another User Group or select Phone programming (see "Day and Night Classes of Service" on **page 400**) and remove COS 2 as a toll restriction.

Chapter 6

TRUNKS AND GATEWAYS

INTRODUCTION

This chapter describes how to program trunks and gateways for your system. A trunk is a communication line between two switching systems. In this guide, the communication line is either the connection between the MiVoice Office 250 Public Branch Exchange (PBX) and the Central Office (CO), or it is the communication line between Mitel PBXs.

Your system may use a SIP or Media Gateway Control Protocol (MGCP) gateway, a device that serves as an entrance and exit into a communications network, to connect trunks to the CO or other networked systems.

This chapter contains the following information:

For information about Service Provider SIP Trunks, see "Service Provider SIP Trunks and SIP Trunk Groups" on **page 691**.

TRUNK AND SPAN TYPES

The MiVoice Office 250 supports the following types of trunks and spans:

- **(Analog) Loop start trunks on the Loop Start Module (LSM-2 or LSM-4):** Trunks that seize lines by bridging through a resistance of the Tip and Ring wires of the telephone line when the phone goes off-hook. Analog Loop start trunks support incoming and outgoing calls and the Caller ID feature. For details, see **page 162**.

- **T1 spans on the T1/E1/PRI (T1M) or Dual T1/E1/PRI (T1M-2) modules:** Digital lines with a signaling speed of 1.544 megabits per second (Mbps) pulse code modulation (PCM) that connect the CO to the phone system. The platform supports the following types of trunks on T1 spans:

  - **Direct Inward Dialing (DID) [Direct Dial Inward (DDI)]**: Digital trunks on a T1 module that can dial an extension directly without going through the attendant. Supports DID, ANI, and DNIS call routing. For information about DID, ANI, and DNIS, see **page 168**.

  - **Ear and Mouth (E&M)**: A (digital) trunk that uses separate leads, called the "E" and "M" leads, for signaling and supervisory purposes. Supports DID, ANI, and DNIS call routing as well as basic networking, such as ring-in to DISA and connection of two systems. For information about E&M trunks, see **page 170**.

  - **Ground Start**: Digital trunks on a T1 module in which signaling occurs when one side of the two-wire trunk, typically the "Ring" conductor of the Tip and Ring, is momentarily grounded to get dialtone. Supports incoming and outgoing calls.

NOTE	1. T1 ground start trunks do not support Mid Call Features (MCF).
	2. T1 ground start trunks do not support Caller ID.

  - **Digital Loop Start**: Supports incoming and outgoing calls and Caller ID.

  - **OPX**: Supports incoming and outgoing calls. OPX circuits are treated as stations and emulate single line phones. See **page 189** for details.

  For additional information about T1 trunks see "T1/E1 Spans and PRI" on **page 162**.

- **Primary Rate T1 trunks on the T1M or T1M-2 module:** A digital transmission system that clusters 24 T1 channels, 23 information-bearing (B) channels and one data (D) channel, for signaling and control. The MiVoice Office 250 supports ISDN Primary Rate trunks on up to three (CS Controller) or four (HX Controller) T1M or T1M-2 modules.
  In Europe and Australia Primary Rate trunks consists of 30 B-channels and 2 D-channels on an E1 line.

- **Basic Rate Interface trunks on the Basic Rate module (BRM-2)**: The Basic Rate Module (BRM-2) provides the Basic Rate 2-Interface ISDN service for communication. The module fully implements the ISDN S/T interface. Each of the two BRI ports consists of two bearer (B) channels and one data (D) channel with HDLC support. The ports provide full I.430 ITU S/T ISDN support for trunks (TE mode). The BRM-2 does not support stations (NT) mode. Each BRM–2 provides two circuits for connecting BRI trunks to the system.

NOTE	The Basic Rate Module (BRM-2) supports only the trunk side of BRI capability. The BRM-2 does *not* support BRI phones or other BRI station-side phones.

  The BRM-2 module is supported only in the Mitel European and Australian markets.

- **Session Initiation Protocol (SIP)**: Trunks that allow the system to use Voice-over-IP (VoIP) outside the network by using the same connection as the Internet connection. See **page 171** for details.

LOOP START TRUNKS

The basis of Plain Old Telephone Service (POTS), loop start trunks signal the Central Office (CO) when a user takes a telephone handset off-hook. Each Loop Start Module (LSM-2) provides two circuits for connecting loop start trunks to a MiVoice Office 250.

The four-port Loop Start Module (LSM-4) provides four circuits for connecting loop start trunks to the MiVoice Office 250.

For detailed information about installing an LSM-2 or LSM-4, refer to the "Installation" chapter in the *MiVoice Office 250 Installation and Administration Manual* .

T1/E1 SPANS AND PRI

This section describes T1 and E1 multi-channel spans and the trunks that can be installed on the MiVoice Office 250 using T1M or T1M-2 modules.

NOTE	This feature requires a single-port or dual-port T1/E1/PRI module (T1M or T1M-2). The second port on the T1M-2 and the PRI feature require software licenses.

T1 Spans

The term "T1" refers to a specific digital method of transmitting voice and data; it is the basic 24-channel time-division multiplex (TDM), pulse code modulation (PCM) technology used in the United States. Since each T1 span actually consists of 24 individual circuits (or channels) multiplexed together, it is often less expensive to purchase a single T1 span than it is to purchase multiple individual trunks.

T1 spans can be programmed to provide the following services:

- **Loop Start Interface**: Loop-start T1 trunks can provide the same services as regular loop-start trunks, including Caller-ID.

- **Direct Inward Dialing (DID) Interface**: Direct inward dialing (DID) allows an outside party to dial into the system without attendant intervention.

- **Ground Start Interface**: Supports incoming and outgoing calls. T1 ground start trunks do not support Caller ID.

NOTE	1. T1 ground start trunks do not support Mid Call Features (MCF).
	2. T1 ground start trunks do not support Caller ID.

- **E&M Interface**: E&M trunks, also called tie lines, are special trunks that tie two distant telephone systems together. They allow the users of either telephone system access to the users and resources of the other telephone system.

- **Off-Premises Extensions (OPX)**: Off-premises extensions are remote single line DTMF phones connected to the T1M or T1M-2 module through telephone company OPX circuits or customer-provided circuits.

Fractional T1 (FT1), which uses fewer than the standard 24 circuits, can be used on the Mitel system. If FT1 is used, the unused circuits of the T1M or T1M-2 port must be unequipped.

Table 27 shows examples of T1 installations.

Table 27:   T-1 Span Applications

System-to-CO T1 #1 —— 6000' max. —— CO T1 #2 —— 6000' max. —— CO	In this application, the T1M or T1M-2 modules are connected directly to the central office. The Line Build Out (LBO) is programmed to support the distance to the CO or the nearest repeater. Both modules draw reference clock from the public network and are slave clocks. One module is designated as the system reference clock.
System-to-System with repeater(s) Master System T1 — 6000' max. — T1 repeater — 6000' max. — Slave System T1	In this application, two Mitel systems are connected through a T1 repeater. Both T1M or T1M-2 modules have the LBO programmed to support the distance to the T1 repeater. One module is designated as the master clock and the other is the slave clock, which receives reference from the master clock.
System-to-System without repeaters Master System T1 — 6000' max. — Slave System T1	In this application, two Mitel systems are connected directly to each other. The T1M or T1M-2 modules have the LBO programmed to support the distance between the Mitel systems. One module is designated as the master clock and the other is the slave clock, which receives reference from the master clock.
Sample T1 Network System #1 T1 #1 — CO 1 System #2 T1 #2 ———— T1 #1 T1 #2 — CO 2	In a T1 network that involves multiple CO and T1 module connections, the LBO is determined by each connection. In the example shown here, the two T1 modules connected to the CO receive reference clock from the public network and are slave clocks. The two that are connecting the Mitel systems do not draw reference from the public network. T1 #2 on System #1 is the master clock and T1 #1 on System #2 is a slave clock.

Dual T1/E1/PRI Module Features

The T1M-2 provides the same functional capability as the T1M. Equipped with two independent digital trunk interfaces, the T1M-2 also provides a dedicated local processor, echo cancellation options, and DTMF tone detection.

The local processor offloads real-time PRI functions from the Processor Module (PM-1). It addresses interrupt latency issues and services the ISDN stack in a timely manner.

Integrated span-side and system-side echo cancellers on the T1M-2 support improved audio quality. The span-side echo canceller addresses field problems when a span-side system is generating echo. The system-side echo canceller works to eliminate echo that may be transported over a T1, T1/PRI, or E1/PRI span from an on-board device.

Nonblocking DTMF tone detection supports a scalable architecture that allows system resources to increase as trunk density increases.

For module installation procedures, refer to the "Installation" chapter in the *MiVoice Office 250 Installation and Administration Manual* .

Variable T1 Span Characteristics

When purchasing a T1 span, several variables are involved, depending on the T1 application required. For details, see **page 162**. The following variables must be considered when ordering a T1 span:

- **Module framing type:** The type of framing scheme used by the T1 spans connected to the module can be D4 Superframe—normally used for voice transmissions—or Extended Superframe (ESF). ESF is usually used for data transmissions.

- **Zero code suppression scheme:** The T1 span zero-suppression scheme, which limits the number of consecutive zeroes in transmissions, for the trunks on the T1M or T1M-2 module can be AMI (Bit 7), Bipolar Eight Zero Substitution (B8ZS), or None. The T1M or T1M-2 module supports B8ZS or None.

- **ISDN switch type:** *(Used for PRI-equipped ports only.)* The system supports the following switch types: AT&T 4ESS and 5ESS Custom, AT&T National ISDN 2, Private Networking, and IP Private Networking.

- **Line build-out (LBO):** The LBO attenuation of the T1 span connected to the port must be designated in programming. This value is determined by the distance to the nearest public network T1 repeater.

- **Reference clock programming:** If the T1M or T1M-2 port is connected to the public network, the port should be designated as a slave clock. The public network always acts as the master clock and one digital trunk port provides the system reference clock. If the module is not connected to the public network, but is instead connected to another T1/EI module or a module in another telephone system, it can be a master clock or a slave clock in relation to the circuit on the other end.

- **Timers:** *(Timers are programmed only for PRI-equipped ports.)* The ISDN timer default values have been carefully selected to ensure proper system operation under most circumstances. Occasionally, one or more of the timers may need to be adjusted.

- **CO Provides Progress Tones**: *(Used for PRI-equipped ports only)* When the system provides local ringback, some central offices (particularly in the UK) take too long to send the system the connect message when a call is received. When the CO connects a call too slowly, the user can answer before the caller is on the line, causing the caller to miss the greeting (such as "Welcome to Mitel. How may I help you?"). If the CO Provides Progress Tones flag is enabled, the system does not provide local ringback. Instead, the system connects the call to the line and the caller hears CO ringback. That way, when the called party answers, the calling party is already on the line. The flag is *disabled* by default.

NOTE	This flag should be enabled for PRI spans that are used for placing outgoing calls from another node (for example, Node 2 uses ARS to place outgoing calls using the PRI span on Node 1). If it is not enabled, some of the calls from Node 2 may not go out to the public network properly. It is recommended that you enable this flag for *all* nodes in a system.

- **Connect On Call Proceeding**: *(Used for PRI-equipped ports only)* This flag affects outgoing Primary Rate ISDN calls. If it is enabled, the system will connect the B-channel as soon

as the Call Proceeding message is received from the CO interface. Some CO interfaces, especially SS7 interfaces, do not send progress indicators when they are playing tones or announcements on the B-channel. They play the tone or announcement prior to sending the progress message, after the call proceeding. In this case, if the flag is disabled, the user experiences audio clipping. If the flag is enabled, the channel is opened when the call proceeding message is received and the system does not have to wait for a progress indicator (indicating alerting, busy, reorder tones or announcements).

- **Operator System Access**: *(Used for PRI-equipped ports set for National ISDN2 only)* When National ISDN 2 is used, there is an option to enable Operator System Access (OSA). OSA is required by some central offices. If this flag is enabled, the user will be allowed to request access to an operator services system.

> **NOTE** If using ARS with National ISDN 2, and OSA is enabled, you must remove Dial Rule #2 Echo Toll Field from the Operator and International Operator Route groups.

- **Send International Toll Digits**: If Enabled, which is the default state of this flag, the system sends international toll digits such as "011" in the dial string in addition to the number type. If Disabled, the toll digits are stripped from the dial string for international calls using ISDN trunks. If the CO ignores the number type, make sure this flag is Enabled.

When ordering Primary Rate Interface services, see **Table 28** for the parameter settings and other information your carrier will need to know:

Table 28:   Primary Rate Interface Parameters and Settings

PARAMETER	SETTING
Protocol	Custom ISDN or National ISDN 2
Signaling	Clear Channel 64Kbs
Framing	ESF
Coding	B8ZS
Jack Type	RJ-48C
Data	Normal (not inverted)
Glare	CPE Yields on Incoming Calls

PRIMARY RATE B-CHANNEL TRUNKS

The MiVoice Office 250 supports Primary Rate B-channels. Each port on a T1M or T1M-2 module consists of 23 B-channels. Each B-channel is programmed as an individual trunk in the database. In Europe and Australia, Primary Rate trunks consists of 30 B-channels and 2 D-channels on an E1 line.

The system is designed to support only AT&T 4ESS Custom, AT&T 5ESS Custom, National ISDN 2, and DMS-100 switch types or Private Networking. No other switch types are currently supported by the Mitel system.

> **NOTE** Primary Rate trunks that are not subject to toll restriction will not return dial tone. This is because these unrestricted trunks are treated as if they are connected to another Mitel system. Therefore, no dial tone is provided, no digits are collected, and the setup is immediate.

B-channel trunks can be combined with other types of trunks in any trunk group.

If a user attempts to place an outgoing call using an individual B-channel trunk, the system sends a Setup Request to the network that requests that B-channel. If the network specifies a different B-channel (possibly because of a glare condition), the system releases the call and the user hears busy tone. The call camps on to the selected B-channel trunk. If the same situation occurs when the user selects a B-channel through a trunk group or ARS, the system will simply move to the next available trunk instead of camping on.

The CO Hookflash [Recall] feature *cannot* be used on B-channel trunks.

Available ISDN services that are supported and not supported by the system appear in **Table 29**. If the B-channel trunks provide a service that is not supported, the system will continue to function properly. However, the system will not make use of the service.

Table 29:   ISDN Services Supported and Not Supported

SUPPORTED BY THE MITEL SYSTEM	NOT SUPPORTED
• Automatic Number Identification (ANI)	• D-Channel Backup
• On-Demand B-Channel Selection	• Flexible Billing
• Dialed Number Identification Service (DNIS)	• Network Ring Again
• Calling Party Number	• Non-Facility Associated Signaling (NFAS)
• Call-By-Call Service Selection	• Primary Rate Inter face Nodal/Centrex Express
	• PRI and Private Line Integration
	• Switched Digital Services
	• User-to-User Information
	• Wide Area ACD

Primary Rate Call Type: All Primary rate ISDN calls have a specific number type (International, National, Network, or Local/Subscriber) and numbering plan field (ISDN, Telephony, or Private) indicated in the call setup message to the public network peer. In certain cases, front end equipment interfacing with the ISDN user (CPE) peer requires a specific number type and/or numbering plan. The number type and numbering plan are programmable per call type for each equipped primary rate circuit. The operation of this feature is transparent to the user. Once the number type and numbering plan are programmed, the Primary rate call setup message will include this information for each call. To set the Primary Rate Call Type for a T1M or T1M-2 module, select System and Chassis, then double-click on the module (or right-click and select Explore Module). Then select Call Type.

Trunk Group PRI Call By Call Feature

System trunk groups provide limited access to PRI Call By Call services. Each trunk group that contains B-channels can be assigned a PRI Call By Call feature to use for outgoing calls. The supported Call By Call features include switched digital circuit services, foreign exchange, TIE services, local exchange, OUTWATS, interexchange carrier services, and custom AT&T and Nortel private networks, features, and services. If the customer wants to use more than one PRI Call By Call feature, there can be multiple trunk groups, each programmed to use a different feature. The customer then can select the Call By Call feature by selecting the trunk group associated with the feature.

When the trunk group has a programmed Call By Call feature, all channels within the specified trunk group will indicate the PRI Call By Call feature code in the outgoing ISDN message. Each outgoing ISDN call sends a SETUP message on the Primary Rate D-channel that contains the B-channel (voice channel) assignment, called digits information, and the network-specific facility. The PRI Call By Call feature is specified within the network-specific facility, ISDN information element which contains the Call By Call feature code.

To properly program the PRI Call By Call feature, the installer must analyze the customer's ISDN outgoing call traffic and configure the amount of B-channels needed to support a particular Call By Call feature or service. In the service order, the installer must request the Call By Call service for the PRI span or a subset of channels.

The installer selects the Call By Call feature during trunk group programming by selecting the desired PRI Call By Call feature. After the trunk group has been created and assigned to a Call By Call feature, the appropriate ISDN B-channels are programmed into the trunk group.

The phone user dials ARS or a PRI trunk group extension. The system selects the trunk group's programmed Call By Call feature. The system automatically inserts the Call By Call feature into the ISDN call control message and transmits the message to the public network. The Public Network accepts or rejects the requested Call By Call feature in the ISDN message. The PRI Call By Call feature supports the following services:

- Call By Call Inactive
- AT&T ACCUNET Switched Digital
- AT&T Banded OUTWATS
- AT&T Carrier Operator
- AT&T DIAL-IT 900 / MultiQ
- AT&T Electronic Tandem Network
- AT&T Foreign Exchange
- AT&T International 800
- AT&T International LDS
- AT&T INWATS
- AT&T Local Operator
- AT&T MEGACOM
- AT&T MEGACOM 800
- AT&T National ISDN INWATS
- AT&T Private Virtual Network
- AT&T TIE Trunk
- AT&T Unbanded OUTWATS
- AT&T WATS Band
- AT&T WATS Maximal Band
- Nortel Foreign Exchange
- Nortel INWATS
- Nortel OUTWATS

- Nortel Private Network

- Nortel TIE trunk

Overlap Sending and Receiving

With overlap sending, the system can send some or no called number digits in the setup message and send additional digits (or overflow digits) in subsequent information messages. You can also program the system to use the overlap sending protocol immediately. If this option is selected, the setup message contains no called number digits, and all digits are sent in subsequent information messages as the digits are dialed.

NOTE	Currently, the overlap sending/receiving protocols are supported with PRI Net 5 and BRI Net 3 switch types, which are prevalent in the European and Mexico markets.

With overlap receiving, the system can receive some or no called number digits in the setup message, followed by overflow digits in information messages. The system will not route the call based on these digits until it receives an indication from the network provider that all digits have been sent. If the network provider does not send such an indication, the system waits until the Overlap Receiving Timeout Timer has expired. Once this timer has expired, the call is routed based on the digits received.

DID/DDI TRUNKS

Direct Inward Dialing (DID) [Direct DIaling Inward, DDI, in Europe] is available on T1M or T1M-2 modules and Single-Line Adapter (SLA) interfaces. For more information about T1 modules, refer to the "Installation" chapter in the *MiVoice Office 250 Installation and Administration Manual* . E&M trunks on T1M or T1M-2 modules can also be designated as DID [DDI] trunks.

DID [DDI] allows an outside party to dial into the system without attendant intervention. To gain direct access to the system, an outside party dials a number that was assigned by the telephone company to the DID [DDI] trunks installed on the system. The system then provides ring signal to the proper phone(s), hunt group, or DISA according to the programmed ring-in for that DID [DDI] number.

DID [DDI] trunks are purchased with blocks of numbers. When a caller dials one of the numbers, the central office rings the trunk that contains that number. When the system answers the call, it handshakes with the central office and receives digits that tell it which DID [DDI] number was dialed. The system then routes the call according to the call routing programmed for that number.

When a call is placed to a DID [DDI] trunk, one of three things will happen:

- If the receiving DID [DDI] trunk is programmed for "Immediate" start type, it will immediately begin receiving the dialed digits to the receiving system.

- If programmed for "Wink" start type, the DID [DDI] trunk performs a "handshake" to signal that it is ready to receive the incoming digits. (This is the default start type.)

- If programmed for "Delay Dial" start type, the DID [DDI] trunk will perform a handshake and pause before receiving any digits.

Each DID [DDI] trunk is programmed with a base number and is programmed to collect a set number of digits. The base number is made up of the digits that are dialed by the caller that are not re-dialed by the central office after the handshake. For example, if the dialed number

is 961-9000 and the base number is 9619, the central office would send only 000 after the handshake. The system would then send the call to the ring-in destination associated with 000 in the trunk's call routing table. The number sent to the phone display could combine the base number and the collected digits to show 961-9000 or can be programmed to show a name.

DID trunks are programmed into trunk groups. However, because DID trunks do not allow outgoing calls to be placed, the outgoing access programming for the trunk group is ignored. If a user attempts to seize a DID/DDI trunk for an outgoing call, the user will hear reorder tones and the phone display will show OUTGOING ACCESS DENIED.

The DID/E&M Receive Busy Instead Of Camp-On phone flag determines whether E&M and DID/DDI callers will receive busy signal or receive ringback and camp on when calling a busy phone. In the default state, busy tones are disabled and the callers hear ringback while camped on to the called phone. This flag is programmed on a phone-by-phone basis. For programming instructions, see "Phone/Phantom/Hot Desk Profile Flags" on **page 417**.
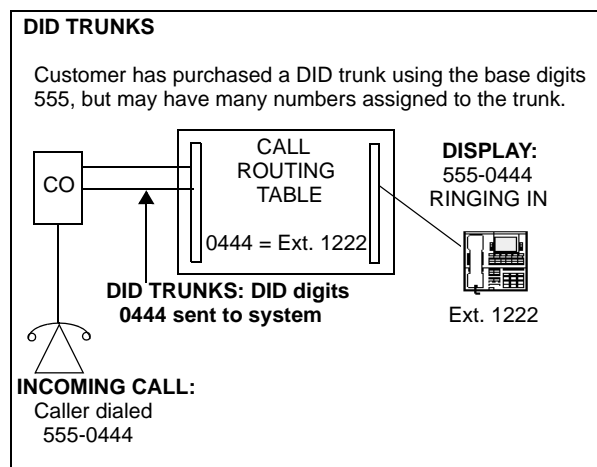
The following programming is required to use the DID [DDI] feature:

1. If installing DID on a T1M or T1M-2 module, do the following:

    a. Install digital trunks. For more information about digital trunks, refer to the "Installation" chapter in the *MiVoice Office 250 Installation and Administration Manual* .

    b. Install and configure a T1M module. For more information about T1 modules, refer to "Installation" chapter in the *MiVoice Office 250 Installation and Administration Manual* . Configure the appropriate circuits as "DID" and assign extension numbers to the trunks. If you are using an E&M trunk, see **page 170**.

    c. Program the T1M or T1M-2 module as required.

2. Program the individual trunks as described on **page 236**.

3. Create a trunk group that contains the DID [DDI] trunks. Program the trunk group to ring in to a Call Routing Table. (Write down the Call Routing Table number. It will be programmed in step 5.) If there are only DID [DDI] trunks in the trunk group, or the trunks will be used only for incoming calls, you can flag the trunk groups as "Incoming Only."

4. Program a Call Routing Table to be used by the DID [DDI] trunk group. Set up the table using the DID [DDI] numbers to match the programming performed in step 3. For more information regarding Call Routing Tables, see **page 248**.

5. If the default state has been changed, enable call information for phone displays. For more information, see **page 174**.

E&M TRUNKS

E&M trunks are special trunks that tie two distant telephone systems together. They allow the users of either telephone system access to the users and resources of the other telephone system. For details about programming overlap sending and receiving, see the previous section.

> **NOTE** This feature is available only on T1M or T1M-2 modules.

E&M trunks, like standard trunks, may be programmed to ring in to call routing tables, individual phones, multiple phones, or hunt groups, or as a DISA call. If programmed to ring-in at a single phone, the E&M call will follow any programmed forward.

When a call is placed over an E&M interface, one of the following events happens:

- If programmed for Immediate start type, the calling system immediately begins sending the dialed digits to the receiving system.

- If programmed for Wink start type, the systems perform a handshake to allow the receiving system to signal that it is ready to receive the digits dialed by the other system. Wink start is the default type.

- If programmed for Delay Dial start type, the calling system waits until its E&M Dial Delay timer expires before sending any digits to the receiving system.

- If programmed for Dial Tone start type, the calling system waits until it receives dial tone from the other system before sending any digits.

E&M trunks can be programmed to support DI. Other options include DNIS and ANI. E&M trunks are programmed into trunk groups. The trunk group programming determines day and night mode outgoing access for the trunks.

When the system is in the default configuration, all E&M trunks are configured for DTMF signaling. If necessary, some or all of the trunks can be reprogrammed for dial-pulse signaling through database programming.

The DID/E&M Receive Busy Instead Of Camp-On phone flag determines whether E&M and DID callers will receive busy signal or receive ringback and camp on when calling a busy phone. In the default state, busy tones are disabled and the callers will hear ringback while camped on to the called phone.

The following programming is required to use the E&M feature:

1. Install and configure a T1M module. For more information about T1M modules, refer to the "Installation" chapter in the *MiVoice Office 250 Installation and Administration Manual* . Configure the appropriate circuits as "E&M" and assign extension numbers to the trunks.

2. Program the T1M or T1M-2 module as required.

3. Program a Call Routing Table to be used by the E&M trunk group. If the Table will be used for DID service, set up the table using the DID numbers. For Call Routing Table programming instructions, see **page 248**.

4. Create a trunk group that contains the E&M trunk(s) and program it to ring in to the Call Routing Table created above. For programming instructions, see **page 190**.

5. Program the individual trunks (see **page 236**). If marked for DID service, program the numbers to match the Call Routing Table created above.

BRI TRUNKS

The Basic Rate Module (BRM-2) provides the Basic Rate 2-Interface ISDN service for communication. The module fully implements the ISDN S/T interface. Each of the two BRI ports consists of two bearer (B) channels and one data (D) channel with HDLC support. The ports provide full I.430 ITU S/T ISDN support for trunks (TE mode). The BRM-2 does not support stations (NT) mode. Each BRM–2 provides two circuits for connecting BRI trunks to the system. See **page 193** for configuring a BRM.

SIP TRUNKS

The system supports SIP (Session Initiated Protocol) trunks to reach the CO. SIP trunks allow the system to communicate with the CO via SIP-enabled gateways. As the SIP protocol becomes more and more popular, it is important to be able to communicate to SIP gateways in the IP-centric world. For more details, see the following pages:

- SIP Gateways on **page 230**
- Gateway SIP Trunks on **page 232**
- Service Provider SIP Trunks and SIP Trunk Groups on **page 691**

TRUNK FEATURES

This section explains the trunk functions and programmable features. For information about trunk capacities, refer to the "Specifications" chapter in the *MiVoice Office 250 Installation and Administration Manual* .

> ### NOTICE
>
> While this system is designed to be reasonably secure against CO trunk misuse by outside callers, there is no implied warranty that the system is not vulnerable to unauthorized intrusions, toll fraud, or unintended toll charges. If the central office does not provide supervision, it will not disconnect the call when one party hangs up, making it possible for a caller to remain connected to a CO trunk circuit. If this happens, and the outside caller remains connected, toll charges could continue to accrue until the outside caller hangs up. Or, if this happens, and the outside caller begins dialing, the call could be placed through the system and would then be billed to the system's owner. The system cannot check this type of call for toll restriction and may not register the call in SMDR. This problem could arise when a call is connected to a phone or when a call is in an unsupervised conference.

TRUNK ACCESS CODES

Use the following feature codes when selecting a trunk to place an outgoing call.

Table 30: Trunk Access Codes

FEATURE NAME	CODE USA [EUROPEAN/ AUSTRALIAN]	DEFINITION
Automatic Route Selection (ARS)	92000	Allows the system to select the desired route for placing a call, as programmed in the database.
Trunk Group Access 1–208	92001–92208	Selects an available trunk from a programmed group of trunks for placing an outside call.
Emergency Calls	911 [999] [112]	Entering this feature code selects an outgoing trunk and automatically dials the programmed emergency number (defaults to Trunk Group 1 and 911 in U.S. systems; 999 in systems in the European market primarily in the UK; 112 in Europe outside of the UK, and Australia).
Outgoing Call	8	Selects an outgoing trunk according to the programmed outgoing access mode for that phone. (Mitel phones use the OUTGOING button or a CALL button.) This code is also used in the European and Australian markets.

ANSWER FEATURE CODE

For calls that are ringing or holding at the phone, the user may enter the Answer feature code (351) or press the **ANSWER** button. When more than one call is ringing or holding, the following priorities determine which call is answered first:

- Ringing calls (ring ins, recalls, callbacks, or transfers) are answered in the order they were received.

- Calls on individual hold are answered in the order they were placed on hold.

CALLER ID, DNIS, AND ANI

The MiVoice Office 250 supports Automatic Numbering Identification (ANI), Caller ID for multi-line phones, and Dialed Number Identification Service (DNIS).

For Release 6.1 and higher, MiVoice Office 250 allows the decoding of Calling Line Identification (CLID) information in countries which use variations of BT CLIP (British Telecom Calling Line Identification Presentation) signaling on analog trunks.

BT CLIP supports the delivery of Caller ID information on analog trunks in the UK, and the standard is commonly known as BT CLIP. BT CLIP is an on-hook capability that provides the user with information about the caller before actually answering a call.

A new trunk Service Type and timer parameters are added in MiVoice Office 250 Database Programming to support BT CLIP. BT CLIP replaces ETSI DTMF Caller ID. When configuring

a Loop Start trunk, the Service Type options now include Bellcore FSK Caller ID and British Telecom FSK.

When configuring a Loop Start trunk, the following Service Type options are supported:

- Bellcore FSK Caller ID

- British Telecom FSK

> **NOTICE**
>
> Your use of Mitel products and/or services, including but not limited to certain Caller ID features, may be subject to US, UK, and Canadian regulations and laws. For example, transmitting misleading or inaccurate caller ID information with the intent to defraud or deceive may be a violation of the Truth in Caller ID Act of 2010. Mitel does not promote use of its products and/or services in such a manner and any such use by you which knowingly contravenes local or national governments laws is at your own risk. Mitel strictly disclaims any liability which may be associated with your wrongful use.

If the Loop Start trunk has been configured with one of the Service Type options above, and caller ID does not work after it has been enabled in Database Programming, perform the following:

1. Check the **Connected to CO** flag. The value should be **Yes**.

2. Run a Hybrid Balance Test (see **page 1121**).

3. Restart Database Programming.

4. Review the results of the test. If the Echo Return Loss (ERL) value is not the optimal setting value, change it to the optimal value (see **page 1124** and **page 1125** for details).

Definitions

The following features provide information about the source of the call.

- **ANI**: Identifies the caller's telephone number. The system receives a specified number of digits.

- ***ANI***: A form of ANI that does not have a specified number of digits. The system receives a star (*) before the ANI digits to signal the beginning of the caller's telephone number. Another star after the digits signals the end of the ANI information.

- **Caller ID**: Provides the caller's telephone number and/or name on multiline phones only.

- **DNIS**: Identifies the number that was dialed to reach your location. The system receives a base number and a specified number of digits that identify the dialed number.

- ***DNIS***: A form of DNIS that does not have a specified number of digits. The system receives a star (*) before the DNIS digits to signal the beginning of the dialed number. Another star after the digits signals the end of the DNIS information.

- ***ANI*DNIS***: A two-stage address service that provides both the caller's telephone number and the dialed number. It combines the *ANI* and the *DNIS* features.

The network retains trunk and outside party information and passes that information along when a call is routed from node to node. With the preceding features and call routing tables activated, the system can identify each incoming call and send it to selected destinations. For example, calls identified with Caller ID or ANI from specific regions of the country can be sent to the

appropriate sales representative or calls from specific locations can be sent to selected individuals. Or, using DNIS information, responses to advertisements using one 800 number can be sent to one hunt group and calls from other 800 numbers can be sent to other hunt groups. Or, they can all be sent to the same destination with different identifying names for the phone displays (such as Magazine Ad, TV Ad, and so forth).

Call information services (Caller ID, ANI, and DNIS) must be enabled for the individual trunks. Loop start trunks can use Caller ID. T1 spans programmed for E&M or DID can use ANI and DNIS. For more information, see "Unsuppressed SMDR Fields by Trunk Service Type" on **page 81**.

The system supports the transmission of Caller ID to single line phones that terminate on IP Single-Line Adapters. This feature uses Automatic Number Identification (ANI), or Caller ID. Once programmed, on-hook single line phones display the calling party's telephone number and name (if available) when receiving an incoming central office (CO) call. The Caller ID information is also displayed if the single line phone receives a transferred call from another phone that has calling party information. Caller ID is *not* transmitted to single line phones attached to Single-Line Adapters (SLAs). However, Caller ID is transmitted to single line phones connected to SLM-4 and SLM-8 ports. All SLM-8 module ports have Caller Identification transmit (CID Tx) capability. Caller ID is transmitted to IP display phones and displayed after the first ring.

Phone Displays

Whether call information appears on the phone display, and what information appears, is determined by phone flags in the database. The flags are:

- **Expanded CO Call Information On Displays:** This phone flag determines whether call information (trunk name or call information) is displayed at the phone. If it is enabled, the Outside Call Party Information Has Priority flag determines what is displayed. If it is disabled, the programmed trunk group user name will appear on the display. In the default state, this flag is *enabled.*

- **Outside Party Call Information Has Priority:** *(not used on single line phones)* If the Expanded CO Call Information flag is enabled, this flag determines what information is displayed at the phone. If enabled, any call that is received on a trunk that provides outside call information (for example, Caller ID or ANI), will be identified on the phone display with the call information. If disabled, the display will show the DID or DNIS information for the call (if available). In the default state, is it *enabled.*

- **Display Outside Name:** If the Display Outside Name phone flag is enabled, the user can switch between the outside party name and number when connected to a CO call with outside party information. The user enters the Display Outside Party Name On/Off feature code (379). In addition, the enhanced ring-in displays will provide the user with more information such as both Caller ID name and number if available, or tell the user if a Caller ID number is blocked or out-of-area. This is a phone-only flag. In the default state, this flag is enabled.

The system will provide the user with advanced displays for direct ring-in calls, if the phone has "Expanded CO Call Information On Displays" and "Outside Party Call Information Has Priority" phone flags enabled (by default they are enabled).

A typical direct ring in display would look like the following: "TRNK GRP 1 RINGING IN." However, with the enhanced displays, the system will attempt to display the name of the outside caller on line 1 and the number of the outside caller on line 2.

When a CO call rings into the system, the system uses the following criteria to determine the name that will appear on the display phones. The criteria are listed from highest priority to lowest priority.

Display Line 1

The following criteria are used to display the name on the top line of the ring-in display. This assumes that all of the necessary phone flags are properly set and usernames are correctly filled in or left blank (at the programmer's discretion) for Call Routing Tables and Trunk Group Names.

1. Outside Party Name provided by the Desktop Application, if available.

2. Outside Party Name provided by Caller ID, if enabled at the phone level.

3. Outside Party Name provided by System Speed Dial. If the collected number matches a number in a Speed Dial bin, the system uses the name of the matching Speed Dial bin.

4. Outside Party Number provided by the Desktop Application.

5. Outside Party Number provided by Caller ID, if enabled at the phone level.

6. Outside Party Number provided by ANI service, if enabled at the phone level.

7. Call Routing Table Name, if the name is not blank.

8. Trunk Group Name, if the name is not blank.

9. Default Trunk Group Name (TG XXXXX).

Display Line 2

The following criteria are used to display the number on the bottom line of the ring-in display. This assumes that all of the necessary flags are properly set.

1. Outside Party Number provided by the Desktop Application, if available.

2. Outside Party Number provided by Caller ID, if enabled at the phone level.

3. Outside Party Number provided by ANI service, if enabled at the phone level.

4. Number Absence Reason, such as Caller ID "OUT-OF-AREA" or "PRIVATE" (blocked) message.

5. RINGING IN display.

ANI and Caller ID information is also sent to the Voice Mail application to provide the caller's telephone number as part of the message envelope.

Caller Information features require the following special programming:

• Caller information display phone flags must be enabled in database programming.

• The call information services must be enabled for the individual trunks.

CALLER ID PROPAGATION

Caller ID (CID) propagation allows Call Processing (CP) to send caller ID information from any incoming services to the public or private network using Primary Rate Interface (PRI) or Basic Rate Interface (BRI) Integrated Services Digital Network (ISDN) circuits. In the European market, CID is known as Calling Line Identification [CLID]. When CID is enabled, the system includes information that identifies the caller to the public or private network.

The CID/CLID information can be a phone username or extension, caller number (Calling Party Number or Name), or incoming CID information. The system software can process any incoming caller ID information and then re-send this information via an ISDN circuit with some limitations. Basically, caller ID received via an ISDN, analog line, or T1 circuit can be processed and sent out to an ISDN circuit. CID information can also be propagated between nodes.

CID information presented by the system results from any one of the following events:

*   A call that is originated from a phone on the system.

*   A call that is from the redirection of an external call.

*   The system programmable fields in DB Programming:

    *   Send Station Extension/Username to Attached PBX on **page 522** (for trunk groups)

    *   Propagate Original Caller ID on **page 523** (for trunk groups)

    *   Do Not Propagate Original Caller ID to P-Asserted Identity on **page 524**

    *   Wait for ISDN Caller ID Information on **page 525** (for trunk groups)

    *   Calling Party Name on **page 411** (for phones)

    *   Propagate Original Caller ID on Transfer on **page 422** (for phones)

    *   Propagate Original Caller ID on Transfer on **page 919** (for voice mail)

See **page 176** for troubleshooting information.

Troubleshooting Caller ID Propagation

The following table lists troubleshooting information for the Caller ID Propagation feature.

Table 31:   Caller ID Propagation Support Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Incoming trunk caller ID does not propagate to an outbound PSTN call.	The original incoming trunk call did not have caller ID.	Enable service with the Central Office (CO) provider to ensure incoming trunk calls have caller ID.
	The "Propagate Original Caller ID for Unanswered Calls" flag for the correct CO trunk group is not enabled.	Enable the "Propagate Original Caller ID for Unanswered Calls" flag for the correct CO trunk group.
	The outbound call is sent out on a non-ISDN PSTN line.	For Caller ID Propagation, the customer must use an ISDN line to propagate caller ID.
	The original trunk is answered by the system (either through a Voice Mail application, human operator, or Unified Communicator).	Currently, the system does not support propagating caller ID for incoming trunk calls that are answered and then transferred back out to the PSTN.
	If the cause is none of the above, the Central Office may not support the ISDN message for propagating caller ID.	Call the CO provider to determine why the ISDN caller ID information is not propagated. If the CO cannot resolve this issue, contact Mitel Technical Support and provide the following information: 1) the ISDN switch type where outbound calls were made; and 2) freeze of this call scenario.

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Table 31: Caller ID Propagation Support Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Caller ID does not propagate to external PBX.	The "Send Station Extension/ Username to Attached PBX" flag is not enabled for the correct CO trunk group.	Enable the "Send Station Extension/ Username to Attached PBX" flag for the correct CO trunk group.
	The outbound call is sent out on a non-ISDN PSTN line.	For Caller ID Propagation, the customer must use an ISDN line to propagate caller ID.
	An unsupported SIP gateway is used to connect the system to an external PBX.	Verify that the SIP gateway has been officially tested. If the SIP gateway is officially supported, call Mitel Technical Support and provide the following information:
		1. the ISDN switch type where outbound calls were made;
		2. the type of gateway used to connect to external PBX;
		3. ISDN switch type settings on SIP gateway; and,
		4. freeze of this call scenario.
		For a more current list of compatible SIP gateways, refer to KB article 09-4940-00056 on the Mitel Knowledge Base Center (http://domino1.mitel.com/ prodsupp/prodsupkb.nsf/ WebSearchForm?OpenForm). Note that you must first log on to Mitel OnLine (http://portal.mitel.com/wps/myportal/ MOLHome) in order to access the KB.
The customer reports station caller ID name (and only name) is not propagating to an external PBX. The customer is using a SIP gateway to communicate to an external PBX.	Two specific conditions cause this to happen: 1) the outbound ISDN line is configured as a DMS-100; and 2) the originating extension is calling out ARS and ARS is routing the call to a remote node.	There is a scenario where station Caller ID Propagation is used over a networked ARS configuration, and the station username will not be passed in the ISDN calling party name field. If the outgoing ISDN line uses the "display IE" method to transfer the calling party name, the calling party name will not use the station's username. Instead of the username, the system uses the calling party name as the name.

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CALLER ID FORWARDING

The following table lists troubleshooting information for the Caller ID Forwarding feature.

Table 32:   Caller ID Forwarding Support Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
The customer reports incoming trunk caller ID does not forward to an outbound PSTN call.	The original incoming trunk call did not have caller ID.	Enable service with the CO provider to ensure incoming trunk calls have caller ID.
	The system administrator has not enabled the "Propagate Original Caller ID for Unanswered Calls" flag for the correct CO trunk group.	Enable the "Propagate Original Caller ID for Unanswered Calls" flag for the correct CO trunk group.
	The outbound call is sent out a non-ISDN PSTN line.	For Caller ID Forwarding, the customer must use an ISDN line to forward caller ID.
	The original trunk is answered by the system (either through a Voice Mail application, human operator, or Unified Communicator).	Currently, the system does not support propagating caller ID for incoming trunk calls that are answered and then transferred back out to the PSTN.
	If the answer is none of the above, the CO may not support the ISDN message for propagating caller ID.	Call the CO provider to determine why the ISDN caller ID information is not forwarded. If the CO cannot resolve this issue, contact Technical Support and provide the following information: 1) the ISDN switch type where outbound calls was made; and 2) freeze of this call scenario.
The customer reports phone caller ID does not forward to external PBX.	The system administrator has not enabled the "Send Station Extension/Username to Attached PBX" flag for the correct CO trunk group.	Enable the "Send Station Extension/Username to Attached PBX" flag for the correct CO trunk group.
	The outbound call is sent out a non-ISDN PSTN line.	For Caller ID Forwarding, the customer must use an ISDN line to forward caller ID.
	An unsupported SIP gateway is used to connect the system to an external PBX.	Verify that the SIP gateway has been officially tested. If the SIP gateway is officially supported, call Technical Support and provide the following information: 1) the ISDN switch type where outbound calls was made, 2) type of gateway used to connect to external PBX, 3) ISDN switch type settings on SIP gateway, and 4) freeze of this call scenario. For a more current list of compatible SIP gateways, refer to KB article 09-4940-00056 on the Mitel Knowledge Base (KB) Center (http://domino1.mitel.com/ prodsupp/prodsupkb.nsf/ WebSearchForm?OpenForm). Note that you must first log on to Mitel OnLine (http://portal.mitel.com/ wps/myportal/MOLHome) in order to access the KB.

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Table 32: Caller ID Forwarding Support Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
The customer reports phone caller ID name (and only name) is not propagating to an external PBX. The customer is using a SIP gateway to communicate to an external PBX.	Two specific conditions cause this to happen: 1) the outbound ISDN line is configured as a DMS-100; and 2) the originating extension is calling out ARS and ARS is routing the call to a remote node.	There is a scenario where phone Caller ID Forwarding is used over a networked ARS configuration, and the phone user name will not be passed in the ISDN calling party name field. If the outgoing ISDN line uses the "display IE" method to transfer the calling party name, the calling party name will not use the phone's user name. Instead of the user name, the system uses the calling party name as the name.

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DAY AND NIGHT MODES

Separate lists exist in the database for class-of-service, outgoing-access, allowed-answer, emergency outgoing access, and ring-in assignments for day and night modes. When an administrator enters the Night Ring feature code, the system uses the night mode lists. When the system is in day mode, the day lists are used. Active calls are not affected when the mode is changed.

Administrators can place any or all remote nodes into night mode or day mode. The default feature code for Enable Network Night is 9861. The default feature code for Enable Network Day is 9862. For more information, refer to the *MiVoice Office 250 Phone Administrator Guide,*.

NOTE	The Night Ring On/Off feature code (9860) affects only the node on which the administrator resides.

DIRECT INWARD SYSTEM ACCESS

Direct Inward System Access (DISA) is a programmable feature that allows an outside party to dial into the system from an external DTMF telephone and then dial extension numbers, hunt group pilot numbers, and off-node device extensions. DISA callers do not have access to outgoing trunks or page zones.

Any of the trunk groups can be programmed to receive incoming DISA calls in day or night mode. When not in use for DISA, the trunk group can be used for placing outgoing calls by phones with outgoing access permission.

NOTE	DISA is not supported on SIP trunks.

Due to the natural characteristics of the trunk, the volume level of DTMF tones transmitted over the trunk may be substantially reduced before reaching the system. This natural degradation in tone volume may adversely affect the reliability of the DISA feature. Other factors which can affect DISA performance are trunk noise and the quality and strength of the DTMF tones generated by the off premises phone itself. If the system cannot recognize a DTMF digit, the call is automatically sent to the primary attendant.

When a DISA user calls a phone extension number, the call rings as a direct ring-in call, even if the called phone is busy or in Do-Not-Disturb mode. The DISA caller hears music or ringing (determined by a system-wide option in database programming) until the Transfer Available or

Transfer Busy timer expires. Then, if the call is not answered, it recalls the called phone's attendant. If the called phone is forwarded, the call follows the programmed forward.

When a hunt group pilot number receives a call through DISA, the call rings or circulates according to how the hunt group is programmed; that is, linear or distributed. If a caller dials a valid hunt group pilot number that has no members assigned to it, the call automatically rings at the primary attendant phone until the Abandoned Call timer expires. If the call is not answered before the Abandoned Call timer expires, the call is disconnected.

Security Codes

DISA trunks can be assigned security codes of up to eight digits that are required for access to the system. The installer can program separate codes for each DISA trunk group to be used during day or night modes.

| NOTE | To prevent unauthorized access to the outgoing trunks, *all* trunk groups using DISA should have a security code. |

System administrators can determine the number of times that a caller may unsuccessfully attempt to enter a security code and/or dial an extension number. If the user does not succeed within the determined number of attempts, the call is disconnected if the security code is invalid or, if the extension number is invalid, the call is transferred to the primary attendant.

Using DISA

| NOTE | If DTMF decoders are unavailable when a DISA call is received, the incoming DISA call is automatically sent to the primary attendant. |

To use DISA:

From a DTMF telephone, the caller dials the telephone number of the DISA trunk. When the call is answered by the system, the caller hears system intercom dial tone
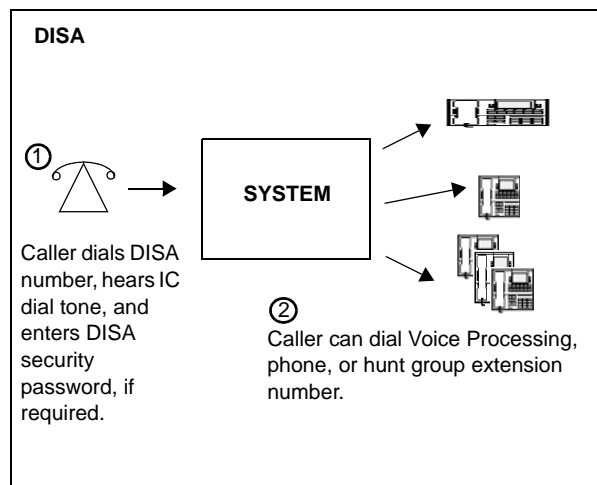
If a single progress tone is heard, the caller must enter the appropriate day or night DISA security code, followed by the pound sign (#). caller dials an extension number or a hunt group pilot number. To call the primary attendant, the caller dials 0. The caller will hear ringing, or may hear music if the system is equipped with Music-On-Hold, until the call is answered or routed in accordance with system programming.

DTMF OR DIAL-PULSE SIGNALING

Trunks can be designated for dual-tone multi-frequency (DTMF) signaling only through database programming.

EMERGENCY CALLS

The MiVoice Office 250 allows immediate access to local emergency facilities whenever a phone user enters the Emergency Call feature code. The dialing pattern defaults to:

- 911 on systems located in the USA

- 999 on systems located in Europe

- 112 on systems located in Australia

When activated, the Emergency Call feature selects a trunk or routes the call based on the phone programming. When activated, the Emergency Call feature overrides all toll restrictions and trunk access programming.

WARNING

Responsibility for Regulatory Compliance.

It is the responsibility of the organization and person(s) performing the installation and maintenance of Mitel Advanced Communications Platforms to know and comply with all regulations required for ensuring Emergency Outgoing Access at the location of both the main system and any remote communication phones. Remote IP and SIP phones may require gateway access to nearby emergency responders.

Emergency Call phone numbers include:

- 911, the default for Mitel systems located in the U.S.
- 999, the default for Mitel systems located in the European market and used primarily in the U.K.
- If applicable, 112, an emergency number used widely in Europe outside of the UK, and Australia.
- Any emergency number, such as for a police or fire station, that is appropriate for the location of the main system and/or remote phones.

Emergency Extensions can be programmed for each phone, and Day/Night Emergency Outgoing Access lists are used to validate the extension making the emergency call.

When a user dials the Emergency Call phone number at intercom or CO dial tone, emergency outgoing access is granted based on system programming and what the user dialed, as described in the following scenarios:

- *If the user dials the emergency feature code from a phone with a trunk or trunk group programmed as the Emergency Extension,* the Emergency Call feature routes the call based on the phone's Emergency Extension. The trunk or trunk group does *not* validate the phone originating the emergency call against the trunk group's Emergency Day/Night Outgoing Access List. An idle trunk is seized, and the system automatically dials Emergency Number 1.

- *If the user dials the emergency feature code from a phone with ARS programmed as the Emergency Extension*:

    a. The Emergency Call feature routes the call based on the phone's Emergency Extension.

   b. The call is routed to Route Group 1.

   c. The call tries the first facility group in Route Group 1.

   d. The trunk group or node trunk group validates the phone originating the emergency call against the group's Emergency Day/Night Outgoing Access List.

   e. *If the call is denied,* it tries the next member in the facility group. If each member denies the call, the call tries the next facility group in the Route Group 1 list.

   If everything is denied due to Emergency Outgoing Access, the call is routed once again to Route Group 1. The call then tries the first facility group in Route Group 1. The trunk group or node trunk group does *not* validate the phone originating the emergency call against the group's Emergency Day/Night Outgoing Access List. If no trunks are available, the call tries the next member in the facility group. If all of the members are unavailable, the call camps-on to Route Group 1 until a trunk is available.

- *If the user accesses a trunk or trunk group and dials any Emergency Number (1–10),* the trunk or trunk group does *not* validate the phone originating the emergency call against the trunk group's Emergency Day/Night Outgoing Access List. An idle trunk is seized, and the system automatically dials the emergency number.

- If the user accesses ARS and dials any Emergency Number (1–10):

   a. The call is routed to Route Group 1.

   b. The call tries the first facility group in Route Group 1.

   c. The trunk group or node trunk group validates the phone originating the emergency call against the group's Emergency Day/Night Outgoing Access List.

   d. *If the call is denied,* the call tries the next member in the facility group. If each member denies the call, the call tries the next facility group in the Route Group 1 list.

   If everything is denied due to Emergency Outgoing Access, the call is routed once again to Route Group 1. The call then tries the first facility group in Route Group 1. The trunk group or node trunk group does *not* validate the phone originating the emergency call against the group's Emergency Day/Night Outgoing Access List. If no trunks are available, the call tries the next member in the facility group. If all of the members are unavailable, the call camps-on to Route Group 1 until a trunk is available.

NOTE	Mitel recommends that only local trunks be installed and used for emergency trunk access and that only local trunks be programmed in Route Group 1. Programming a node in Route Group 1 may cause the system to access a trunk on a separate node when ARS is used for emergency access. If this occurs, Emergency Outgoing Access is no longer validated.

The system allows the Dialing Wait After Connect timer to expire and then dials the digit string programmed in the database as the emergency number.

At the time the call is processed, a minor alarm will be generated by the system and sent to all administrator phones. Also, if the Message Print option is enabled, the alarm message is sent

to the designated output port (remote node).

```
┌─────────────────────────────────────────────────────────────┐
│  EMERGENCY CALL FEATURE                                      │
│                                          CALL PLACED         │
│                                          TO 911              │
│                    ┌──────────────────┐                      │
│                    │ SYSTEM DATABASE: │                      │
│                    │ EMERGENCY EXT. = │                      │
│   [phone]   ───►   │ TRUNK GROUP 1 (92001)                   │
│                    │        OR        │──►  ALARM #11         │
│                    │ DIRECT TRUNK ACCESS│    DISPLAYED        │
│  Phone user        │ EMERGENCY No. = 911│                    │
│  dials 911         └──────────────────┘                      │
│                                          [printer]           │
│                                                              │
│                    MESSAGE PRINTOUT TO REMOTE                │
│                             NODE, IF ENABLED                 │
└─────────────────────────────────────────────────────────────┘
```
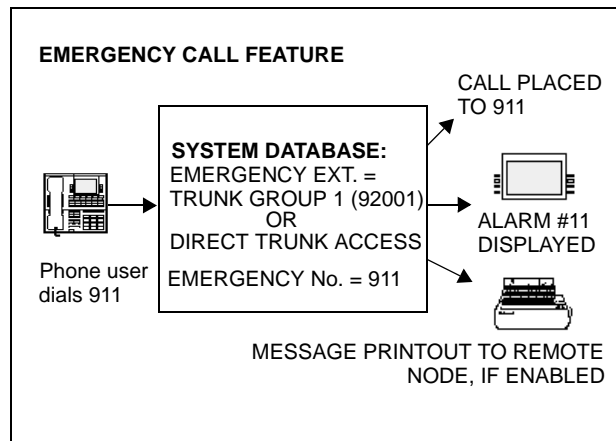
Emergency Outgoing Access Lists

The system supports two outgoing access lists called *Emergency Day Outgoing Access* and *Emergency Night Outgoing Access*. These lists, which are similar to the Day/Night Outgoing Access lists, are programmable for CO Trunk Groups and Node IP Connection Groups. These lists default to the Auto Extension List *PP051: Auto: All Phones*. For programming instructions, see "Extension Lists and System Groups" on **page 499**.

Ten Emergency Numbers

The system can store up to 10 emergency numbers that are dialed when the Emergency Call feature is used.

If the system is installed in an area where emergency responder services such as 911 in the USA, 999 in the European market, or 112 in Australia are not available, Mitel recommends substituting the phone number for the local police or fire department or the telephone company operator.

This feature can be made inoperative by removing all trunk access and/or by removing the dialed digit string. If Emergency Call is not operational, the system presents a warning message that the feature has been disabled. The programmer must acknowledge or change this condition before system operation can continue.

Like other feature codes, the default Emergency Call feature code can be changed from 911, 999, or 112 to a different code, if necessary.

An emergency number will be blocked if there are no trunks or emergency numbers programed in the database, or all trunks in a trunk group are busy.

If the Emergency Call feature is programmed to use ARS, route group 1 (local calls) will be used even if its dial patterns are reprogrammed.

Emergency calls, by default, use the first local trunk group and will not be sent using node trunk groups on other nodes. However, when ARS is used to place an emergency call, Route Group 1 is used, even if it contains nodes. This means that the network *can* access a trunk on a node other than the user's node if the user accesses ARS and dials the emergency number. *Local*

trunks must be installed and used for emergency number trunk access and nodes should not be used in Route Group 1. When a user places an emergency call, every administrator in the network receives an emergency alarm.

Calling Party Number Field

Each phone can be programmed to send an identifying number when a call is placed. This information is required by government regulation for emergency calls in some areas. Up to 48 digits can be programmed in the Calling Party Number field. However, check with your service provider to determine their specific requirements. This number is sent in the ISDN setup message in the Calling Party Number Information Element. In addition, the system sends the extension number of the phone in the Calling Party Number Subaddress Information Element. The CO should ignore this information element if it does not support the feature.

NOTE	No default number exists for this field. It is up to the system administrator to supply the correct Emergency Calling Party Number for each phone.

If an off-premises extension is used for dialing an Emergency Number such as 911 in the USA, 999 in the European market, or 112 in Australia, the emergency responder operators will see the Calling Party Number or the address the system chassis location rather than the address of the off-premises location. Off-premises personnel should be prepared to give the correct address and other pertinent information if it is not programmed as the Calling Party Number.

For programming instructions, see "Phones and Devices" on **page 353**.

IP Devices and Local Loop Support

The system supports SIP (Session Initiated Protocol) trunks to reach the CO. SIP trunks allow the system to communicate with the CO via SIP-enabled gateways. As the SIP protocol becomes more and more popular, it is important to be able to communicate to SIP gateways in the IP-centric world.

For a more current list of compatible SIP gateways, refer to KB article 09-4940-00056 on the Mitel Knowledge Base Center (http://domino1.mitel.com/prodsupp/prodsupkb.nsf/WebSearchForm?OpenForm). Note that you must first log on to Mitel OnLine (http://portal.mitel.com/wps/myportal/MOLHome) in order to access the KB.

WARNING

Possible Delay in Local Emergency Response to Remote Sites.

IP and SIP phone users should be alerted to the following hazardous situations:

- If an Emergency Call phone number is dialed from an IP or SIP phone located at a remote site that is **not** equipped with a correctly configured gateway, the call will be placed from the location where the system chassis is installed rather than from the location where the emergency call is made.

- In this situation, emergency responders may be dispatched to the wrong location. To minimize the risk of remote site users misdirecting emergency responders, Mitel recommends regular testing of MGCP/SIP gateway trunk(s) for dial tone.

- If uninterruptible power supply (UPS) protection has *not* been installed as part of the MiVoice Office 250, IP and SIP phones will *not* operate when electrical power fails either at remote sites or at the main system location.

To place calls during a power failure in this situation, IP and SIP phone users can only use a single line phone connected to one of the power failure bypass circuits built-in to the system chassis. If a phone connected to a power failure bypass circuit is not available, users should make emergency calls **from a local phone not connected to the system**. For details about installing phones to the Power Failure Bypass circuits, refer to the "Installation" chapter in the *MiVoice Office 250 Installation and Administration Manual* .

When configured with an MGCP Gateway or a SIP Gateway, the system can be programmed so that when an Emergency Number is dialed from an IP phone, the calling name and number associated with the IP phone—not the system—is sent to the emergency response operators. To ensure this functionality works, the system must be programmed for remote loop termination.

NOTE | The MGCP gateway is intended to connect to the PSTN. Connecting the MGCP gateway to single line ports on other systems is not supported.

OUTGOING-ACCESS, ALLOWED-ANSWER, AND RING-IN ASSIGNMENTS

Each trunk group has programmed lists of phones for outgoing-access, allowed-answer, and ring-in assignments for day and night modes.

- **Outgoing-access**: Permits the phone user to place calls using trunks in that trunk group. Each phone has a default outgoing access code programmed in the database. When the user presses the **OUTGOING** button, presses an idle CALL button, or enters the Outgoing Call feature code, 8 (0 in Australia), the system automatically selects an outgoing trunk.

NOTE | Because the network does not allow users to directly access trunks on other nodes, each trunk group's Outgoing Access can contain only phones on the local node. Users must use ARS to access trunks on other nodes.

- **Allowed-answer**: Permits the phone user to answer incoming calls on the trunks in that trunk group (even if the phone does not have ring-in assignment for that trunk group). Phones cannot have allowed-answer assignment for trunk groups on other nodes. On phones, the individual trunk button flashes (if one exists) to indicate the ringing call. If a phone is programmed with allowed Answer Access only (no ring-in) for a trunk group, direct ring-in calls on that trunk group can be answered by entering the Automatic Trunk Answer feature code (350) or pressing the flashing individual trunk button.

- **Ring-in**: Assigns ring-in destinations which can be one or more phones, modems, extension lists, or Voice Processing applications; to a hunt group; to DISA; or to a call routing table.

A trunk can ring in to a phone or Voice Processing application on another node. The ring in destination(s) for the trunk group will receive direct ring-in calls on trunks in that trunk group. On phones with ring in, the individual trunk or CALL button flashes and the display indicates a ringing call. Allowed answer for the trunk group is automatically assigned to a phone with ring in.

When phones have outgoing-access or ring-in assignments, the associated individual trunk buttons show the status of their trunks.

Phones that do not appear on any of the lists cannot place or directly receive outside calls; they are limited to intercom calls, conferences, transferred calls, and retrieving calls on system hold. (A call on system hold can only be picked up at the phone that placed it on hold or at a phone that has an individual trunk button and has allowed-answer and/or outgoing access for that trunk.)

A private trunk group with one or more trunks can be established by programming outgoing-access, ring-in, and allowed-answer permission for the trunk group to only one phone.

For programming instructions, see "Extension Lists and System Groups" on **page 499**.

TRUNK MANAGEMENT

This section describes the way the MiVoice Office 250 has been designed to facilitate efficient management of trunk facilities.

TRUNK GROUPS

Each trunk is assigned to a trunk group. Trunk group feature codes and trunk group buttons are used to select a trunk in one of the programmed trunk groups. Each trunk must be assigned to a trunk group. For example, all local trunks could be in one group, while another group could contain WATS trunks that are used for long distance calling. Unused trunks can be placed in a single trunk group that is labeled "unused."

All trunks in a trunk group must reside on the same node as the trunk group. The trunk group is programmed to process outgoing calls in linear or distributed order:

*   **Linear trunk group**: Requests for an outgoing trunk are always processed beginning with the highest numbered trunk circuit on the list and move through the list until an available trunk is found.

*   **Distributed trunk group**: The first request will be processed beginning with the highest numbered trunk circuit on the list. The next request will begin with the second highest numbered trunk, and each subsequent request will begin one trunk lower on the list. When the end of the list is reached, requests begin again with the highest numbered trunk on the list.

The Camp On feature can be enabled or disabled for each trunk group. If the Camp-on feature is disabled, callers will hear continuous busy signals when all trunks in the trunk group are in use or unavailable. If Camp On is enabled, callers are able to camp on to the trunk group and wait for an available trunk. See **page 526** for details about trunk Camp-Ons.

IP GATEWAY TRUNKS

IP gateway trunks terminate on a SIP gateway in order to interconnect the MiVoice Office 250 to the public switch telephone network (PSTN). SIP gateway trunks can be programmed on the MiVoice Office 250 without having an adequate number of IP gateway trunk licenses. However, in this situation the system allows only the licensed number of IP gateway trunks to come online, based on Call Processing's IP gateway trunk order.

All SIP gateway trunks come online first, based on their hardware addresses, whether or not a physical connection to the SIP gateway exists. This means the IPR Application uses the first available IP networking resource to bring the SIP gateway trunks online.

From the end-user point of view, hardware addresses are not used. Internally, however, Call Processing uses hardware addresses.

In online mode, DB Programming posts a warning if the user attempts to program more IP gateway trunks than the number licensed.

The system allows only the number of licensed IP gateway trunks to come online. When more that the licensed number attempt to come online, the Message Print log identifies the trunk extensions that failed to come online. At the same time, Alarm #127 appears on the unit chassis' LCD panel and on the Administrator phone display.

NODES AND NODE TRUNK GROUPS

Node trunk groups: Are made up of PRI circuits that are programmed for private network use. When a PRI-equipped module is programmed for networking (Private Networking switch type), the system automatically creates a node trunk group. (All B-Channels residing on the same T1M or T1M-2 module are in the same node trunk group.)

Node programming: Contains a list of node trunk groups used to access another node. For each node in the network, you must define the routes to every other node. For example, in a network with four nodes, you would define three routes for each node (one to each of the other three nodes). For more information about private networking refer to "Appendix A: Private Networking," in the *MiVoice Office 250 Installation and Administration Manual* .

The nodes are programmed to select their node trunk groups in linear or distributed order:

- **Linear**: The node first attempts to route through the first node trunk group listed in the node. If the node is unable to route through that node trunk group, it attempts to route through the second node trunk group listed in the route. The node continues to attempt to route through subsequent node trunk groups in the list until it successfully routes or exhausts all node trunk groups in the list.

- **Distributed**: The node shifts the first node trunk group it attempts to use. For example, if the node routed the previous call or communication through the first node trunk group listed, the node routes the second call or communication through the second node trunk group listed.

To prevent circular or excessive routes, the system limits the number of "hops." A call or communication can take up to 10 hops to other nodes. However, 2 or 3 hops maximum is recommended.

T1 OPX SERVICE

Some sites may experience difficulties when using a loop start T1 line configured for OPX service. This arrangement may be used to connect two Mitel systems in order to perform semi-networking applications such as intercom access, lighting message lamps, and shared voice mail. The reason for the difficulty is that the T1 OPX does not send a disconnect signal to the loop start trunk, which causes trunks to lock-up.

For T1 OPX phones, the flag "Send T1 OPX Disconnect Flash" sends a proprietary disconnect signal from the T1 OPX to the loop start trunk. The "A" bit is toggled high for the duration of the SL Disconnect Flash Duration timer. At default, this flag is disabled. It is programmed on a phone-by-phone basis and is available only to single line phones. See "Send T1 OPX Disconnect Flash" on **page 422** for details.

To use this phone flag, the system must be equipped with a T1M or T1M-2 module.

NOTE	The Send T1 OPX Disconnect Flash flag does not affect T1 channels configured for Loop Start. Therefore, a Mitel system can be on the receiving end with T1 Loop Start channels and will recognize the disconnect. However, it cannot **send** the disconnect unless it has T1 OPX enabled.

MODULES AND PORTS

The Controller programming area allows you to designate the type of interface module that is installed in each of the module bays of the Base Server. In addition, there is information which tells you the number of timeslots required by the database configuration. This is done so that you are aware of the number of timeslots in use by the system.

Although the basic system configuration is non-blocking, certain configurations can lead to circumstances wherein more timeslots are assigned than are available, which may result in blocking if an attempt is made to use all resources at once. For further discussion of time slot allotment, see **"Time Slot Groups"** on **page 868**. In addition to the Processor Module (PM-1), Base Server modules can be installed in any of the three available bays. When the System is an HX Controller, you can program up to four Bays instead of three Bays. The Digital Expansion Interface supports modules that cannot be installed in the Base Server.

The following trunk-related modules are available for installation (see **page 191** for details):

- T1/E1/PRI Module

- Dual T1/E1/PRI Module

- Two-port/four-port Loop Start (LSM-2/LSM-4)

- Two-port Basic Rate Module (BRM-2) [Australian/European markets only]

The following phone-related modules are available for installation (see "Phones and Devices" on **page 353** for details):

- Four-port/eight-port Single Line Module (SLM-4/SLM-8)

NOTE	On the CS Controller chassis, there are two, on board single line ports, instead of a separate module as on the HX Controller.

- Digital Desktop Module-16 (DDM-16)

- Digital Endpoint Module (DEM)

When changing the module type, existing circuits must first be reconfigured. Not redefining the associated circuits before inserting another module type of a different device type produces an error message. The fields used in Controller programming are described on **page 192**.

For information about the Processor Module and Processor Expansion Card, see "Local Processor Module and Expansion Card IP Settings" on **page 617**.

This section contains the following information:

- Base Server Module Bays on **page 191**

- Power Fail Bypass Capabilities on **page 215**

- Programming Echo Thresholds on **page 215**

- Troubleshooting Modules on **page 216**

Bays and Slot Numbers

The hardware address "slot" number is visible when viewing devices equipped on a module in the Bay.

Table 33 summarizes the slot numbers used in the MiVoice Office 250.

Table 33:   Bays and Slot Numbers

BAY	SLOT
Bay 1	1
Bay 2	2
Bay 3	3
Bay 4	15
Processor Module	7
Processor Expansion Card	9
Loop Start Ports	10
Single Line Ports	11
DEI #1 Bay 1	4
DEI #1 Bay 2	5
DEI #1 Bay 3	6
DEI #2 Bay 1	12
DEI #2 Bay 2	13
DEI #2 Bay 3	14
DEI #3 Bay 1	20
DEI #3 Bay 2	21
DEI #3 Bay 3	22
DEI #4 Bay 1	28
DEI #4 Bay 2	29
DEI #4 Bay 3	30

BASE SERVER MODULE BAYS

This section contains the following information:

- **Supported Modules** below

- Moving Modules on **page 193**

- Configuring a Basic Rate Module (BRM-2) on **page 193**

- Configuring a Two-Port/Four-Port Loop Start Module (LSM-2/LSM-4) on **page 197**

- Configuring a Single-Port/Dual-Port T1/E1/PRI Module (T1M/T1M-2) on **page 199**

- Programming the Busy Out Manager on **page 213**

SUPPORTED MODULES

When the System is an HX Controller, you can program up to four Bays instead of three Bays in the Controller folder.

Four module bays support the following modules:

- Basic Rate Module - 2 (BRM-2) [Australian/European markets only]

- T1/E1/PRI Module, supported, but no longer in production

- Dual T1/E1/PRI Module (T1M-2)

- Loop Start Module - 2 ((LSM-2), supported, but no longer in production)

- Loop Start Module - 4 (LSM-4)

- Digital Desktop Module - 16 (DDM-16)*

- Single Line Module - 4 (SLM-4)*

* For information about the DDM-16 and SLM-4, see "Phones and Devices" on **page 353**.

Modules can reside in any bay and in any combination. Module bays displaying "Uninstalled" have not been configured with a module type listed above. If you intend to configure a T1/E1/PRI module for PRI use, you must have the PRI premium feature license, part no. 840.0227.

To select the module type that will be assigned to a particular bay, click the desired bay labeled Uninstalled, then select the desired module type in the list box.
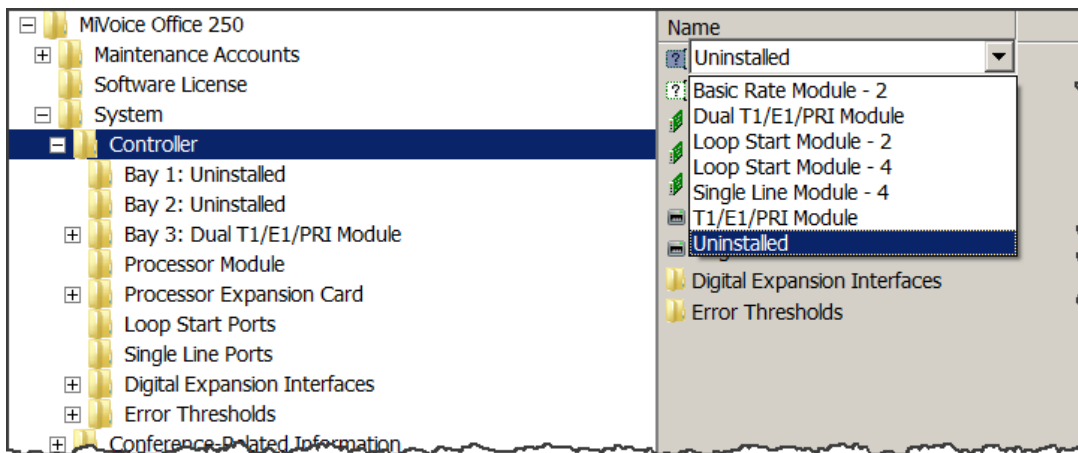
When a module is changed, one of the following may occur:

- When a bay is modified to contain a Loop Start Module or T1/E1/PRI module, the new ports will be included in the "Unused CO" trunk group.

- When a module bay is set to *Uninstalled*, or where the existing module type is changed to another module type, **all** reference to the phones or trunks that were on the module in that bay will be removed from **all** areas of the Database Manager (unless the module was first moved to another bay).

- The bay that contains the module with the primary attendant phone cannot be changed to another module type or deleted until the attendant phone is changed to another location on another module. If attempted, a warning message appears.

To program an individual module, double-click it (or right-click and select **Explore *<module name>***). Follow the programming instructions for that type of module:

- Configuring a Basic Rate Module (BRM-2) on **page 193**

- Configuring a Two-Port/Four-Port Loop Start Module (LSM-2/LSM-4) on **page 197**

- Configuring a Single-Port/Dual-Port T1/E1/PRI Module (T1M/T1M-2) on **page 199**

- Configuring a Four-Port Single Line Module (SLM-4) on **page 373**

If you change the module type, a window appears prompting you to confirm the change. Select **Yes** to continue or **No** to cancel the change.

MOVING MODULES

A module can be moved to any unoccupied location except the left-most bay, which is larger than the others and reserved for the Processor Module.

To move a module, select the existing module and drag and drop it to the new location. Or, select the module to move and press **CTRL** + the up/down arrow to move the module up or down in the list.

You can also move a module using the Move Module shortcut menu option. This procedure can only be done while in local programming mode.

1. Right-click the desired module or press **CTRL + M**.

2. Select **Move Module**. The Move Module dialog box appears.
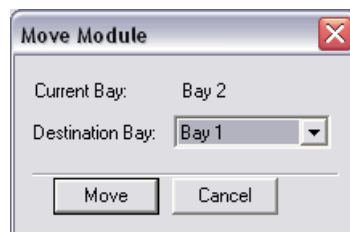
3. Select the destination bay, and then click **Move**.

CONFIGURING A BASIC RATE MODULE (BRM-2)

The Basic Rate Module (BRM-2) provides the Basic Rate 2-Interface ISDN service for communication. The module fully implements the ISDN S/T interface. Each of the two BRI ports consists of two bearer (B) channels and one data (D) channel with HDLC support. The ports provide full I.430 ITU S/T ISDN support for trunks (TE mode). The BRM-2 does not support stations (NT) mode. Each BRM–2 provides two circuits for connecting BRI trunks to the system.

To change a specific port, double-click **Ports**. A Basic Rate Module - 2 can also be configured in the Configuration Wizard. See "Launching the Configuration Wizard" on **page 59** for

instructions on how to launch the wizard. For complete information about the Configuration Wizard, refer to the *MiVoice Office 250 DB Programming Help*.

> **NOTE** Basic Rate Interface (BRI) capability applies to Australian and European market installations for this release.

To equip several ports at once:

1. Select the ports you want to equip.

   To select a series of items, hold down **SHIFT** while selecting the first and last item in the range. To select two or more that are not consecutive, hold down **CTRL** while selecting the desired items.

2. Right-click and select **Batch Change Type**. The Batch Change Type dialog box appears.

3. Click the port type you want to assign, and then click **OK**.

BRM programming includes the following fields:

- **Call Type** below

- Overlap Sending on **page 195**

- Port Status on **page 195**

- SPID/DN/TEI on **page 195**

- Timers on **page 196**

- Description on **page 196**

- ISDN Switch Type on **page 197**

- CO Provides Progress Tones on **page 197**

- Connect on Call Proceeding on **page 197**

- Process ISDN Data Link Down on **page 197**

- Retry ARS Call IF Call Rejected on **page 197**

- D-Channel Diagnostics on **page 197**

Call Type

Call Type is a North American-only setting.

All Basic Rate ISDN calls have a specific number type and numbering plan field indicated in the call setup message to the public network peer. In certain cases, front-end equipment interfacing with the ISDN user (CPE) peer requires a specific number type and/or numbering plan. In the past System was unable to support this requirement because the number type and numbering plan options were not programmable. Now the number type and numbering plan are programmable per call type for each equipped primary rate module. The operation of this feature is transparent to the user. Once the number type and numbering plan are programmed, the Primary rate call setup message will include this information for each call.

Overlap Sending

(Applies to BRMTrunk ports) If using a BRM-equipped module with BRM Net 5, you can program the system to use the overlap sending and receiving protocols. These protocols allow you to program the switch to send called number digits either in the setup message or the information message. See "To access the programming options:" on **page 206** for programming details.

Port Status

The screen displays the status of the following:

*   **Out of Sync**: When this field displays a NO value, it indicates that the port is synchronized with the terminal connection. If it is a trunk port, this indicates that the connection with the CO (local exchange) is operational. If it is a phone port, it indicates that the signal is being transmitted to any devices connected to the port. If the value is YES, then the port is not synchronized with its terminal connection, but may still have signal. Further information will be provided in the Troubleshooting section of the Installation and Field Maintenance Manual.

*   **Loss of Signal (LOS)**: This field indicates that the port either does (a value of NO) or does not (a value of YES) recognize the device or CO (local exchange) trunk it is connected to. When the Loss of Signal field has a value of YES, the Out of Sync value should also be YES. When the Loss of Signal field has a value of NO, the Out of Sync field can assume either value and provides additional information about BRI status.

*   **2B Loopback:** When the 2B Loopback Test is active, data entering the device on the B1/B2 receive channels is demodulated and remodulated back out on the B1/B2 transmit channels. Data is received and transmitted on the same B channel, either B1 or B2, and does not cross channels. This test checks the transmission and reception between the BRS Module and the public network, verifying the integrity of the BRS chip transceiver and the physical interface from the module to the public network. BRS phone ports cannot perform loopback tests on the D channel.

SPID/DN/TEI

The SPID/DN/TEI option applies only for BRI trunk ports only. Each BRI trunk can have up to 16 SPID/DN/TEI pairs.

*   **Directory Number (DN)**: (US Only) Is the number. No DN numbers are assigned by default.

*   **Service Profile Identification (SPID)**: (US Only) Contains the bearer information so that the CO can associate the number with a set of bearer types, The CO will reject calls to the directory number based on the bearer types assigned to that number. This also allows the CO to charge a higher rate for numbers that have more bearer capabilities (for example, numbers which support data or video as opposed to just speech). No SPIDs are assigned by default.

*   **Terminal Phone Identifier (TEI)**: Contains value that Network uses to identify the terminal. This value can be fixed or automatically assigned by the Central Office. In a point-to-point setup, there is only one TEI. In a point-to-multipoint setup, there is at least two. For the US, each SPID/DN pair has an associated TEI. The default is set to Automatic.

Some Central Offices do not allow 16 SPIDs, and some allow as few as six. Also, SPID/DN/ TEI pairs are only defined for North American ISDN switch types, so they are defined only for the US Systems. Other countries do not support SPID/DN.

Timers

To program a timer:

1. Select its current Value and enter the new value in the text box.

2. Press **ENTER** or click on another field to save the change.

The timers are:

- **Busy No Answer** (T301): Applicable to system for Japan only. If the Call Delivered response (connect message) is not received before this timer expires, the System disconnects the call.

- **Call Proceeding** (T310): See **page 204**.

- **Connect Acknowledge** (T313): See **page 204**.

- **Data Link Disconnect** (T309): See **page 204**.

- **Disconnect** (T305): See **page 204**.

- **Idle Link** (T203): See **page 204**.

- * **Overlap Receiving** (T302): See **page 204**.

- * **Overlap Sending** (T304): See **page 204**.

- **Release** (T308): See **page 204**.

- **Restart Acknowledge** (T316): See **page 205**.

- **Resume Acknowledge** (T318): See **page 205**.

- **Retransmit TE1 Request UI Message** (T202): Controls the amount of time the ISDN Data Link layer will wait for a response from the network before it re-transmits a UI frame during the TEI assignment procedure.

- **Re-Transmission** (T200): See **page 205**.

- **Setup Request Acknowledge** (T303): See **page 205**.

- **Status Enquiry** (T322): See **page 205**.

- **Suspend Acknowledge** (T319): See **page 205**.

- **XID Request** (T204): See **page 205**.

* Currently, the overlap sending/receiving protocols are supported with PRI Net 5 and BRI Net 3 switch types, which are prevalent in the Europe and Mexico markets.

Description

To enter a description:

1. Select the current Value, then enter the description in the text box.

2. Press **ENTER** or click another field to save the change.

ISDN Switch Type

Basic Rate Module supports AT&T 5ESS Custom, DMS-100, and National ISDN2 switch types.

To change the ISDN switch type setting:

1. Select the current Value, and then use the drop-down list box to scroll to the desired setting.

2. Press **ENTER** or click another field to save the change.

CO Provides Progress Tones

See **page 211**.

Connect on Call Proceeding

See **page 211**.

Process ISDN Data Link Down

This field allows a system to determine whether or not the BRI trunk port processes an ISDN Data Link Down message. When this flag is set to Yes and a BRI trunk port receives an ISDN Data Link Down message, then a BRI trunk port drops all calls and attempts to bring the ISDN data link up. When this flag is set to No and a BRI trunk port receives an ISDN data link down message, then the BRI trunk ignores the ISDN data link down. Some Central Offices (local exchanges in Europe) will release its TEI after a B-channel trunk has been idle for some time. This causes an ISDN data link down message to be sent from the CO (local exchange) to the switch. In this case, if the switch processes the ISDN data link down message, it may drop the call on the other B-channel trunk on that port. The default value of this field is to ignore the ISDN data link down message.

Retry ARS Call IF Call Rejected

See **page 212**.

D-Channel Diagnostics

This field turns on/off D-channel diagnostics for BRM. It is set to Off by default. The selections are: Off, Layer 1, or Layer 2.

To change the setting:

1. Select the current Value, and then use the drop-down list box to scroll to the desired setting.

2. Press **ENTER** or click another field to save the change.

CONFIGURING A TWO-PORT/FOUR-PORT LOOP START MODULE (LSM-2/LSM-4)

The system provides two or four loop start ports. The ports provided in these modules are in addition to the built-in ports of the same type. The primary difference is that the built-in ports also provide the Power-Fail Bypass functionality.

When the System is an HX Controller, the on-board Loop Start capacity is expanded from two to four ports. DB Programming shows four onboard Loop Start ports. Although the boards are set up as 4-port boards, only the original two ports are equipped by default.

The figure below shows an example of the Loop Start Ports 1-4.
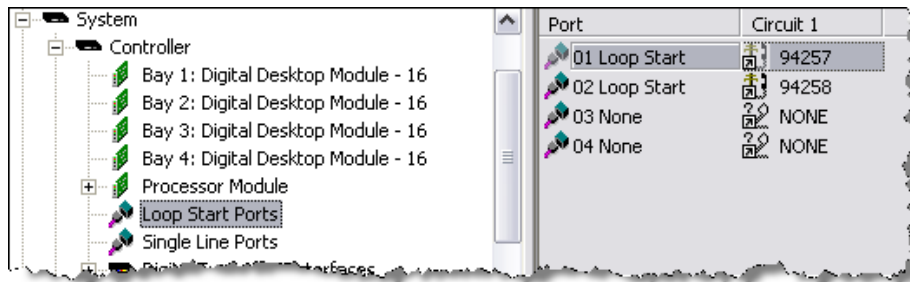
With the introduction of the HX Controller chassis, DB Programming makes all attempts to appropriately show the hardware that is available for both HX and CS systems. If three Loop Start ports are programmed into a database and then the platform is changed to CS Controller, a red X appears over the unavailable third port.

To equip a single port:

1. Select the current Port description and use the drop-down list box to scroll to the new port type.

2. Press **ENTER** or click another field to save the change.

To equip several ports at once:

1. Select the ports you want to equip.

    To select a series of items, hold down **SHIFT** while selecting the first and last item in the range. To select two or more that are not consecutive, hold down **CTRL** while selecting the desired items.

2. Right-click and select **Batch Change Type**. The dialogs below appear.

3. Click the port type you want to assign, and then click **OK**.

Moving a Loop Start Port

When DB Programming is in offline mode, you can drag/drop a port to a module that has available ports ("None"). You cannot move devices in online mode. If you try to drop on a fully programmed module, an error message appears.

To "move" a Loop Start port:

1. Open the desired database in an offline mode session of DB Programming. Make sure the desired module is explored so the "None" port move location is viewable in the left-hand tree view.

2. Browse to the module with the existing port you want to move. Click the port, hold down the mouse button, and drag it over to the tree view. The None port will become highlighted when the port to be moved is dragged onto it.

CONFIGURING A SINGLE-PORT/DUAL-PORT T1/E1/PRI MODULE (T1M/T1M-2)

Depending on the selection, one of the following options can be configured for the T1M module (or up to two of the options for the T1M-2 module):

• T1

• T1/PRI

• E1/PRI

Auto-Equipping/Upgrading from the Single-Port T1/E1/PRI Module

If the T1M-2 module is installed in an uninstalled expansion bay prior to starting an online mode session of DB Programming, the module will auto-equip and be displayed in the appropriate bay of the Controller when the DB Programming session is established.

If a T1M was previously programmed in the bay for the T1M-2 module, you may change the module from the T1/E1/PRI module to the Dual T1/E1/PRI module in DB Programming. The programming for the T1M port will be automatically copied to the first port of the T1M-2. Conversely, if a T1M-2 with two configured/programmed ports is changed to a T1M, the programming for the first port of the T1M-2 will be preserved and configured for the T1M port, while the programming for the second port of the T1M-2 will be deleted.

DB Programming also permits drag/drop of T1M or T1M-2 ports from one T1M/T1M-2 module to any "None" port of a T1M/T1M-2. To preserve programming when transitioning from one module type to another, the ability to "move" a port from one T1M/T1M-2 to another was added.

IMPORTANT	DB Programming will display an error message under the following circumstances: • Change the port type of one port if the other port is programmed (not None). • Change the Framing Type under a T1 port and the other port is programmed (not None). This field also exists under the T1/PRI port type, but it is OLM only. DB Programming displays: WARNING: Changes to this field are only recommended on an idle system as this may cause a short pause in audio on the other port of this module (but no dropped calls). Would you like to continue?

NOTE	The move option is available only in an offline mode session of DB Programming. Only T1, T1/PRI or E1/PRI ports may be moved. The None port may not be moved. Only None ports are valid destinations for a move.

To "move" a T1M/T1M-2 port:

1. Open the desired database in an offline mode session of DB Programming. Make sure the desired module is explored so the "None" port move location is viewable in the left-hand tree view.

2. Browse to the module with the existing port you want to move. Click the port, hold down the mouse button, and drag it over to the tree view. The None port will become highlighted when the port to be moved is dragged onto it.
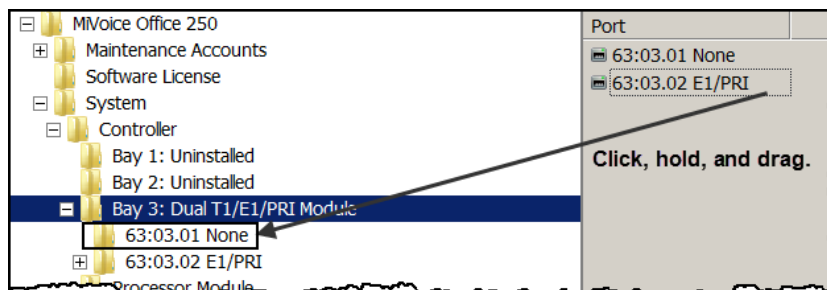
3. Drop the port by releasing the mouse button over the desired None port. The port will be "moved" from the original module to the new module and all of its programming (circuits, networking, and so forth) will be retained. The original module will now have "None" listed as the port type where the moved port used to be.
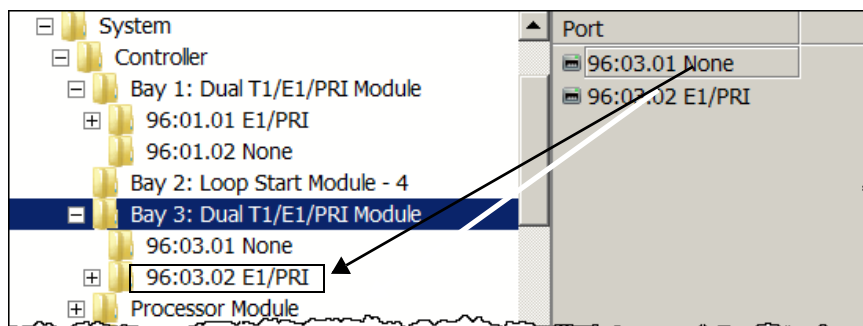
When changing a module type from T1M to T1M-2 or vice versa or drag/dropping a port between module types, the echo profiles for devices programmed on any ports undergoing the module type transition will be changed to their default type. See "Echo Profiles" on **page 735** for default echo profiles based on module and device type. Also, when going from a T1M to a T1M-2 (either changing the module type or drag/dropping a port) the Span Connection Type port-level field of the Dual T1/E1/PRI ports will be defaulted to the system companding type.

To program a port on one of the T1M or T1M-2 modules, double-click the module to display the programming fields. Module Programming includes the following fields.

Table 34:   T1M and T1M-2 Module Programming Fields

FIELD	REF. PAGE	T1	T1/PRI	E1/PRI
T1/E1 Circuits	201	✓	✓	✓
T1/E1 Diagnostics	202	✓	✓	✓
Timers	203		✓	✓
Call Type	205		✓ (T1M-2 only)	
Overlap Sending	206			✓

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Table 34:   T1M and T1M-2 Module Programming Fields (continued)

FIELD	REF. PAGE	T1	T1/PRI	E1/PRI
Description	**207**	✓	✓	✓
Enable On-Module Dialing	**207**	✓	✓	✓ (T1M-2 only)
Framing Type	**208**	✓		
Zero Code Suppression	**208**	✓		✓
Line Impedance	**208**			✓
CRC Processing	**208**			✓
Haul Mode	**209**	✓	✓	
Line Build-Out	**209**	✓	✓	
Loop Length	**210**	✓	✓	
ISDN Switch Type	**210**		✓	✓
CO Provides Progress Tones	**211**		✓	✓
Connect on Call Proceeding	**211**		✓	✓
Enable ISDN Two B-Channel Transfer	**268**		✓	✓
Operator System Access	**211**		✓	✓
Retry ARS Call If Call Rejected	**212**		✓	✓
Send International Toll Digits	**212**		✓	✓
D-Channel Diagnostics	**213**		✓	✓
Always Send Call Proceeding Before Connect Request	**213**		✓	✓
Span Companding Type	**213**	✓ (T1M-2 only)		✓ (T1M-2 only)

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T1/E1 Circuits

To select the port type for a specific circuit:

1. Select the current Circuit Type description and use the drop-down list box to scroll to the new port type.

2. Press **ENTER** or click another field to save the change.

To equip several circuits at once:

1. Select the circuits you want to equip. To select a series of items, hold down the **SHIFT** key while selecting the first and last item in the range. To select two or more that are not consecutive, hold down **CTRL** while selecting the desired items.

2. Right-click and select **Batch Change Type**. One of the dialogs below appears.

<div align="center">

T1 Options: **T1/PRI or E1/PRI Options:**

</div>
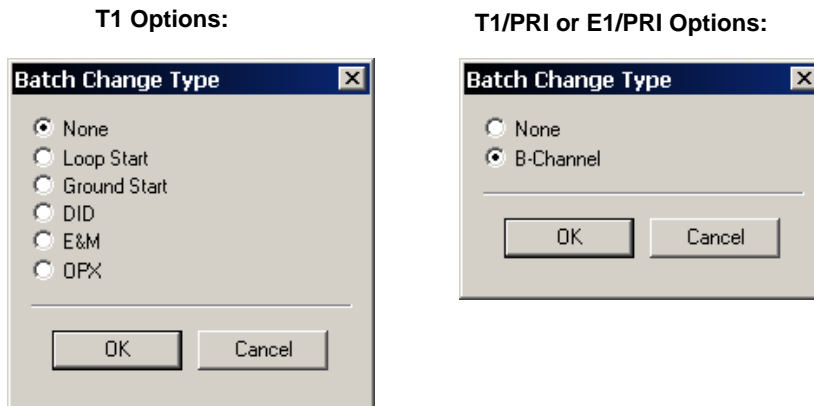
3. Select the circuit type you want to assign, and then click **OK**.

If the new devices require extension numbers, you will be prompted to select the first number in the batch. The available extension numbers will be assigned sequentially to the newly equipped circuits. If you attempt to exceed the device limit of the System, an error message appears informing you the operation cannot be performed.

T1/E1 Diagnostics

T1 Diagnostics is used for troubleshooting T1 or E1. It shows the status of various conditions for the selected module.

- **Hourly Display**: Double-click **Hourly** to see the number of error that have occurred in each hour for the past 24 hours.

- **Daily Display**: Double-click **Daily** to see the number of errors that have occurred during several 24-hour periods.

- **Hourly Summary**: The hourly summary shows the number of errors that have occurred in the past hour.

- **Daily Summary**: The daily summary shows the number of errors that have occurred in the past 24 hours.

- **Last Error Update Hour and Day**: These fields show the date and time that the information was last updated. (Information is automatically updated whenever a new module is selected.)

- **T1/E1 Status**: Double-click **T1/E1 Status** to see the following fields. They show the status of the T1 or E1 module and allow you to perform diagnostics. To update the information, right-click anywhere and select **Update All Statuses**.

  - **Make Busy**: This option can be used to busy-out all of the circuits. When it is enabled, the module is busied out, just as it would be if the busy-out switch on the module was

used. To change this option, right-click the current Value and then select **Enable Make Busy**. To turn off the Make busy, right-click again and select **Disable Make Busy**.

NOTE	The T1 or E1 circuits should be busied out before any diagnostics procedures are performed. If not, the module will go into Red Alarm and drop all active calls.

If the Make Busy LED is flashing rapidly on the module, someone pressed the Make Busy switch. If this occurs, you must disable this Make Busy field.

- **Loopback Tests**: These options can be used for enabling and disabling the following T1 loopback diagnostics and alarm. During the tests, LED indications and error messages can be used to provide troubleshooting information. (The testing options cannot be used in stand-alone programming mode.) To enable or disable one of the tests, right-click its current Value, then select **Enable**. To end the test, right-click again and select **Disable**. If the T1 or E1 Loopback timeout timer expires during a loopback test, the loopback will be terminated automatically. The loopback tests are as follows:

  - *Digital Local Loopback:* Allows you to check the transmission and reception of alarm conditions between systems in a private T1 or E1 network. This tests the framing chip on the module.

  - *Digital Remote Loopback:* Allows you to check the transmission and reception of test patterns conditions generated by the T1/E1/PRI module. This tests the framing chip on the module.

  - *Transceiver Local Loopback:* Allows you to check the transmission and reception of alarm conditions generated by the T1/E1/PRI module. This tests the transceiver chip on the module.

  - *Transceiver Remote Loopback:* Allows you to check the transmission and reception of test patterns between the T1/E1/PRI module and the public network. This tests the transceiver chip on the module.

  - *Transmit Blue Alarm:* This sends a Blue alarm over the public or private network interface for testing purposes.

- **Line Build-Out**: This information is shown here for reference purposes only. To program Line Build-out for the T1/E1/PRI module, see **page 209**.

Timers

(Timers are programmed only for PRI-equipped modules). The ISDN timer default values, listed in Table 28 on **page 165**, have been carefully selected to ensure proper system operation under most circumstances. Occasionally, one or more of the timers may need to be adjusted.

To program a timer:

1. Select the current Value, and then enter the new value in the text box.

2. Press **ENTER** or click another field to save the change.

Table 35:   PRI-Equipped Module Timers

TIMER	DEFAULT	RANGE	PURPOSE
Call Proceeding (T310)	25	1–255 sec	*This is used for AT&T 5ESS custom switch type only.* If the appropriate Call Proceeding response (Alerting, Connect, or Disconnect) is not received before this timer expires, the requesting system sends a Disconnect.
Connect Ack. (T313)	9	1–255 sec	If the appropriate Call Proceeding response (Alerting, Connect, or Disconnect) is not received before this timer expires, the requesting system sends a Disconnect.
Data Link Disconnect (T309)	90	1–255 seconds	*(T1/PRI Only)* This timer is initiated when the data link layer is disconnected. The timer is terminated when the data link is re-established.
Disconnect (T305)	4	1–255 sec	If the appropriate Disconnect response (Disconnect or Release) is not received before this timer expires, the requesting system sends a Release.
Idle Link (T203)	100	1–65535 tenths	This timer controls how long the ISDN Data Link layer will remain idle before transmitting a Receive Ready (RR) Supervisory frame or an information frame, once multi-frame operation has been established.
Overlap Receiving (T302)*	15	1–255 sec	This timer determines the number of seconds that the system will wait between messages during the overlap receiving process. This timer is started whenever the system receives a setup acknowledge message. It is then restarted every time the system receives an information message during the overlap receiving process. Once the last information message is received, the system has until this timer expires to send a call progress message (for example, alerting, connected, and so forth). If the system does not send a call progress message or if the system does not receive an information message before this timer expires, the call is cleared.
Overlap Sending (T304)*	20	1–255 sec	This timer determines the number of seconds that the system will wait between messages during the overlap sending process. This timer is started when the system receives the setup acknowledge message from the network provider. It is then restarted every time the system sends an information message during the overlap sending process. Once the last information message is sent, the system waits for a call progress message (for example, alerting, connected, and so forth) from the network provider. If the system does not receive a call progress message or if the system does not send an information message before this timer expires, the call is cleared.

* Currently, the overlap sending/receiving protocols are supported with PRI Net 5 and BRI Net 3 switch types, which are prevalent in the Europe and Mexico markets.

TIMER	DEFAULT	RANGE	PURPOSE
Release (T308)	4	1–255 sec	If the appropriate Release response (Release Complete) is not received before this timer expires, the requesting system sends another Release or releases the B-channel and Call Reference value.

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Table 35:   PRI-Equipped Module Timers (continued)

TIMER	DEFAULT	RANGE	PURPOSE
Restart Acknowledge (T316)	120	1–255 sec	The system initiates this timer when a Channel Restart message is sent. The timer is terminated when the Restart Acknowledge is received from the public network. If the Restart Acknowledge is not received before this timer expires, the system sends a second Restart message to the public network. If the timer expires a second time and a Restart Acknowledge has not been received from the public network, the system idles the respective channel.
Resume Acknowledge (T318)	9 seconds	1–255 sec	*(E1/PRI Only)* If the appropriate Resume response (Resume Acknowledge or Resume Reject) is not received before this timer expires, the requesting system sends a Release Request.
Re-Transmission (T200)	10	1–65535 tenths	This timer controls the amount of time the ISDN Data Link layer will wait for an acknowledgment from the network before it transmits a frame again.
Setup Request Ack. (T303)	4 (9 for 4ESS)	1–255 sec	If the appropriate Setup Request response (Alerting, Connect, Call Proceeding, Setup Acknowledge, or Release) is not received before this timer expires, the requesting system sends another Setup Request or clears the call.
Suspend Acknowledge (T319)	9 sec	1–255 sec	*(E1/PRI Only)* If the appropriate Suspend response (Disconnect, Release, Suspend Acknowledge, or Suspend Reject) is not received before this timer expires, the requesting system sends a Suspend Reject.
XID Request (T204)	20 tenths	1–65535 tenths	This timer controls how long the ISDN Link Data layer will wait for an XID response from the network before transmitting the XID Request again. (XID parameter negotiation is not supported.)

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Call Type

(Applies only to T1/PRI-equipped modules.) All Primary rate ISDN calls have a specific number type (International, National, Network, or Local/Subscriber) and numbering plan field (ISDN, Telephony, or Private) indicated in the call setup message to the public network peer. In certain cases, front end equipment interfacing with the ISDN user (CPE) peer requires a specific number type and/or numbering plan. The number type and numbering plan are programmable per call type for each equipped primary rate module. The operation of this feature is transparent to the user. Once the number type and numbering plan are programmed, the Primary rate call setup message will include this information for each call.

To set the Primary Rate Call Type for a T1/PRI module:

1.   Click **System**, and then click **Controller**.

2.   Double-click the module (or right-click and select Explore Module).

3.   Select **Call Type**.

Overlap Sending

(Applies only to E1/PRI equipped modules) If using a PRI-equipped module with PRI Net 5, you can program the system to use the overlap sending and receiving protocols. These protocols allow you to program the switch to send called number digits either in the setup message or the information message.

To access the programming options:

1. Double-click **Overlap Sending**.

2. Program either of the following options:

    •   **Immediate Overlap Sending**: Enable this flag to have the system immediately send the setup message to the network provider. If disabled, all called number digits are collected before sending the setup message to the network provider. You should enable this flag if you want the system to seize a B-channel immediately or if you want the network provider to provide immediate feedback for invalid numbers. You should disable this flag if the network provider does not support overlap receiving. By default, this flag is *disabled*.

NOTE	Because Automatic Route Selection (ARS) calls require the system to collect digits before determining the most cost-effective route, this flag is ignored for ARS calls.

    •   **Maximum Digits in Called Number IE**: Select the maximum number of called number digits that the system will include in the setup and/or information messages. This field may be required if the network provider only allows calls with a specified number of maximum digits. The number of digits programmed here and the Overlap Sending Conditions value determine the number of called number digits sent in the called number information elements (IE) for setup and information messages. The valid range is 1–48, and the default is *20*.

NOTE	If the Immediate Overlap Sending flag is enabled, this field is ignored.

    •   **Overlap Receiving**: Enable this flag if the network provider does not send digits "en bloc" to the system. If enabled, the system will not route the call until it has received a sending complete indication from the network provider or until the Overlap Receiving Timeout Timer has expired (below). You should enable this flag if the network provider sends any called number digits in subsequent information messages. You should disable this flag if the network provider sends all of the called number digits in the setup message ("en bloc"). By default, this is *disabled*.

    •   **Overlap Receiving Timeout Timer**: Select the number of seconds that the system will wait to receive any called number digits from the network provider. You may need to adjust this value if the network provider does not send a sending complete indication to signal the end of the called number digits. The valid range is 1–255, and the default is *30* seconds.

    •   **Overlap Sending Conditions**: Select how the system will send digits if the Immediate Overlap Sending flag is disabled. Whether or not the overlap sending protocol is used depends on the value selected here. You have the following options:

- • *Overlap Sending Disabled:* Sends all digits "en bloc" as part of the called number IE in the setup message. This option does not allow overlap sending.

- • *Overflow Is Equal Access Only:* Sends any overflow digits as equal access digits in the setup message. This option does not allow overlap sending. This is the default setting.

- • *Overflow Digits Sent In Overlap:* Sends any overflow digits in subsequent information messages. This option allows overlap sending.

- • *Overlap And Equal Access Facility:* Sends any equal access digits in the setup message, but sends overflow digits that are not considered equal access in subsequent information messages. This option allows overlap sending.

- • *Overlap Sending Enabled For All Calls:* Sends all digits in subsequent information messages (that is, none are included in the setup message). The number of digits sent in each information message is limited by the Maximum Digits in Called Number IE setting. This option allows overlap sending.

- • **Sending Complete Indication**: Select how the system will indicate that it has finished sending digits to the network provider. If Sending Complete IE is selected, the system includes a sending complete information element in the setup or information message. If '#' Digit in Called Number IE is selected, the system inserts a pound (or hash) digit (#) at the end of the digits in the called number information element. By default, this is *Sending Complete IE*.

NOTE	This field is ignored if the Transmit Sending Complete Indication flag is disabled.

- • **Transmit Sending Complete Indication**: Enable this flag to have the system send the complete indication type selected in the Sending Complete Indication field. If this flag is disabled, the system will not send any indication to the network provider that no more digits follow. By default, this is *disabled*.

NOTE	The Transmit Sending Complete Indication and Sending Complete Indication fields also affect ISDN trunks that use the "en bloc" method for delivering digits. For example, if the Transmit Sending Complete Indication flag is enabled and the Sending Complete Indication value is set to '#' Digit in Called Number IE, the system will insert a pound (or hash) digit (#) after the collected digits.

Description

To enter a description:

1. Select the current Value, then enter the description in the text box.

2. Press **ENTER** or click another field to save the change.

Enable On-Module Dialing

In previous versions of the MiVoice Office 250, there was a system level flag labeled Enable T1/E1/PRI Dialing. This flag has been removed and replaced with this port level flag for the T1M and T1M-2 modules. This flag handles pulse and DTMF dialing for T1M/T1M-2 ports. It is set to **Yes** by default.

To change the Enable On-Module Dialing setting:

1. Select the current Value and then place a check mark in the box by selecting it. To remove the check, select the box again.

2. Press **ENTER** or select another field to save the change.

Framing Type

The type of framing scheme used by the T1/E1 trunks connected to the module can be as follows:

- T1 Trunks: D4 Superframe or Extended Superframe (ESF)

- E1 Trunks: Common Channel Signaling or Channel Associated Signaling

To set the framing type:

1. Select the current Value, then use the drop-down list box to scroll to the desired setting.

2. Press **ENTER** or click another field to save the change.

IMPORTANT	DB Programming will display an error message under the following circumstances: • Change the port type of one port if the other port is programmed (not None). • Change the Framing Type under a T1 port and the other port is programmed (not None). This field also exists under the T1/PRI port type, but it is OLM only. DB Programming displays: WARNING: Changes to this field are only recommended on an idle system as this may cause a short pause in audio on the other port of this module (but no dropped calls). Would you like to continue?

Zero Code Suppression

The trunk zero-suppression scheme supports the followings:

- T1/PRI and T1/E1/PRI modules: AMI (Bit 7), Bipolar Eight Zero Substitution (B8ZS), or None

- E1/PRI modules: HDB3 or None

To change the setting:

1. Select the current Value, then use the drop-down list box to scroll to the desired setting.

2. Press **ENTER** or click another field to save the change.

Line Impedance

(Applies only to E1/PRI-equipped modules.) Set this value to the line impedance the CO is using for E1/PRI. The value here must match the CO setting. The system supports the following line impedances:

- 120 ohms (default)

- 75 ohms

CRC Processing

(Applies only to E1/PRI equipped modules.) The CRC Processing flag is a configuration flag that can be enabled/disabled for each E1 module. The setting for this flag must match the

setting for the remote-end E1 signal. When CRC Processing is enabled only at one end of an E1 connection, the E1 signal with the CRC enabled will never reach the CRC Multiframe synchronization state and will transmit the RAI signal to the remote end until either the CRC Processing is disabled for that E1 signal or the CRC Processing is enabled for the remote-end E1 signal.

This flag enables the transmission of a CRC Multiframe pattern and a CRC-4 value computed/ transmitted for every 16 frames of data. At the same time, the CRC Processing flag enables the E1 module to look for this CRC Multiframe pattern and CRC-4 value on the received E1 signal. When CRC Processing is enabled on both ends of an E1 connection, both ends will reach the CRC Multiframe synchronization. In this multiframe synchronized state, both ends will compute a CRC-4 value and transmit it to the other end along with the E1 data. On the receive ends of the E1 connection, both receivers will then calculate their own CRC-4 value for the received data and if the received and the computed CRC-4 values match, the corresponding 16 frames of E1 data are assumed to be have been received with no bit errors.

Haul Mode

This field applies only for a T1 or T1/PRI configuration (E1/PRIs do not display this field). Short-haul implies a shorter span and the corresponding loop length field lists lengths for the cable. Long-haul implies a longer cable with the need for a repeater and the line build out field lists its associated signal strengths. In previous versions, only Long Haul was available for the T1 or T1/PRI. It is set to **Long Haul** by default.

To change the setting:

1. Select the current Value, then use the drop-down list box to scroll to the desired setting.

2. Press **ENTER** or click another field to save the change.

Line Build-Out

(Applies only to T1 and T1/PRI modules.) The LBO attenuation of the T1 trunk connected to the module value is determined by the distance to the nearest public network T1 repeater. The selections are shown below.

- 0 dB (DSX-1)

- −7.5 dB (this is the best setting for T1/E1/PRI modules)

- −15 dB

- −22.5 dB

The Auto option should be used during initial installation. The other settings can be used if the build-out needs to be adjusted later.

To change the setting:

1. Select the current Value, and then use the drop-down list box to scroll to the desired setting.

2. Press **ENTER** or click another field to save the change.

Loop Length

(Applies only to T1 and T1/PRI modules.) This is displayed with a red "X" if the Haul Mode is set to Long Haul. This field provides optional loop lengths pertaining to a Short Haul configuration. The selections are shown below:

- 0-133 ft. (default)

- 133-266 ft.

- 266-399 ft.

- 399-533 ft.

- 533-655 ft.

To change the setting:

1. Select the current Value, and then use the drop-down list box to scroll to the desired setting.

2. Press **ENTER** or click another field to save the change.

ISDN Switch Type

(Applies only to PRI-equipped modules.). The system supports Primary Rate B-channels.

- T1/PRI modules support AT&T 4ESS Custom, AT&T 5ESS Custom, DMS-100 or National ISDN 2, Private Networking, and IP Private Networking.

- E1/PRI module supports Private Networking, PRI NET-5 S-type, and IP Private Networking.

If the module being programmed will be used to connect two nodes in a network, select Private Networking.

| NOTE | **For B-Channel Devices**: When changing the switch type field in the port folder of a T1/PRI or E1/PRI port, any B-Channel devices will be moved to their appropriate default echo profile based on the switch type value. B-Channels are associated with different echo profiles based on which module type they are programmed on and whether they are being used for private networking or as CO trunks. See "Echo Profiles" on **page 735** for the default echo profile table. |

To change the ISDN switch type setting:

1. Select the current Value, and then use the drop-down list box to scroll to the desired setting.

2. Press **ENTER** or click another field to save the change. If you selected Private Networking as the ISDN Switch Type, you will be prompted to select the node that will contain the trunks on this module. If you want to change the node later, use one of the following methods:

   Method A

   a. Select the current Value, then enter the new value in the text box.

   b. Press **ENTER**. A screen appears displaying what is associated with the number.

   c. Click **OK**. The new number appears in the field.

   Method B

a. Right-click the existing Node Number. An option box appears.

b. Select **Change Node Number**. A window appears prompting for the type of device to include.

c. Select **Local Node** or **Remote Node**, and then click **Next**. The list of nodes appears. (You can view a list by selecting List or view details by selecting Details.)

d. Select the node, then click **Finish**. The selection appears as the Node Number.

NOTE	If you change the ISDN Switch Type to Private Networking, the system automatically places the trunks in the first available node trunk group. For more information about node trunk groups, see "Node Trunk Groups" on **page 548**. If you change to any other ISDN Switch Type option, all of the trunks are moved to the "Unused" trunk group.

CO Provides Progress Tones

(BRI/PRI-equipped modules only) When the System provides local ringback, some central offices [local exchanges], particularly in Europe, take too long to send the System the connect message when a call is received. When the CO connects a call too slowly, the user can answer before the caller is on the line, causing the caller to miss the greeting (such as "Welcome to Mitel. How may I help you?"). If this flag is enabled, the System does not provide local ringback. Instead, the System connects the call to the line and the caller hears CO ringback. That way, when the called party answers, the calling party is already on the line. In the default state, the flag is *disabled*.

To change the CO Provides Progress Tones setting:

1. Select the current Value and then place a check mark in the box by selecting it. To remove the check, select the box again.

2. Press **ENTER** or select another field to save the change.

NOTE	This flag should be enabled for PRI spans that are used for placing outgoing calls from another node (for example, Node 2 uses ARS to place outgoing calls using the PRI span on Node 1). If it is not enabled, some of the calls from Node 2 may not go out to the public network properly. It is recommended that you enable this flag for **all** nodes in a system.

Connect On Call Proceeding

(PRI-equipped modules only) This flag affects outgoing Primary Rate ISDN calls. If it is enabled, the system will connect the B-channel as soon as the Call Proceeding message is received from the CO interface. Some CO interfaces, especially SS7 interfaces, do not send progress indicators when they are playing tones or announcements on the B-channel. They play the tone or announcement prior to sending the progress message, after the call proceeding. In this case, if the flag is disabled, the user experiences audio clipping. If the flag is enabled, the channel is opened when the call proceeding message is received and the system does not have to wait for a progress indicator (indicating alerting, busy, reorder tones or announcements).

Operator System Access

(T1 PRI set for National ISDN2 only) If National ISDN 2 is selected, there is an option to enable Operator System Access. Operator System Access (OSA) is required by some central offices. If this flag is enabled, the user will be allowed to request access to an operator services system.

> **NOTE** | If using ARS with National ISDN 2, and OSA is enabled, you must remove Dial Rule #2 Echo Toll Field from the Operator and International Operator Route groups. See "Default Route Groups" on **page 145**.

Retry ARS Call If Call Rejected

The Retry ARS Call If Call Rejected flag is for ISDN messaging for outgoing calls using ARS. If set to "Yes," the system retries the additional channels and trunk groups programmed in ARS if the call is rejected by the Telco when receiving a specific cause value.

The symptom that will appear because of this flag is when an outgoing call is rejected by Telco, where it causes the system (via ARS) to cycle through all the ISDN channels before releasing/disconnecting the call.

This flag must remain enabled or set to "Yes" on T1 or E1 PRI circuits connected to Interprise 3200s. This flag is enabled by default. The flag should be disabled on Public ISDN circuits connected to Telco. If not using ARS with ISDN services, this flag does not apply.

When this flag is set to "No," the system does not retry the ARS call, regardless of the reason it was rejected. If this flag is set to "Yes," the system retries the ARS call as long as the cause of the rejection is not one of the following events:

- Unallocated or unassigned number
- User is busy
- No user is responding
- Call rejected
- Number has changed
- Destination is out of order
- Number format is invalid
- There was a temporary failure
- Congestion in the switching equipment
- Requested circuit/channel is not available
- Outgoing calls are barred
- Incoming calls are barred
- Service or option not available, unspecified
- Service or option not implemented, unspecified
- Mandatory information element is missing
- Interworking, unspecified

Send International Toll Digits

If enabled, the system sends international toll digits (for example, 011) in the dial string in addition to the number type. If disabled, the toll digits are stripped from the dial string for

international calls using ISDN trunks. You should enable this flag if the CO ignores the number type. By default, this feature is *Enabled*, set to Yes.

D-Channel Diagnostics

This field turns on/off D-channel diagnostics for T1/PRI or E1/PRI. This is not displayed for T1. It is set to Off by default. The selections are: Off, Level 1, or Level 2.

To change the setting:

1. Select the current Value, and then use the drop-down list box to scroll to the desired setting.

2. Press **ENTER** or click another field to save the change.

Always Send Call Proceeding Before Connect Request

Setting the Always Send Call Proceeding Before Connect Request flag to Yes forces the system to send a CALL_PROCEEDING message to the network before sending a CONNECT message for incoming ISDN calls, in cases where the system would normally send an immediate CONNECT message in response to an incoming SETUP message. It is set to No by default.

Span Companding Type

This is the companding type of the remote end connected to the port. Defaults to the System Level Companding Type (under System\Flags).

To change the setting:

1. Select the current Value, and then use the drop-down list box to scroll to the desired setting.

2. Press **ENTER** or click another field to save the change.

PROGRAMMING THE BUSY OUT MANAGER

The Busy Out Manager allows you to select ports or circuits from a list and specify Busy Out commands for the selected items. The Busy Out Manager displays all ports on the selected module as well as what type of port is configured (a "None" device will be shown if the port has not been configured). Selecting a port displays the devices/circuits programmed for that particular port (devices/circuits that have not been configured will *not* show up as a None device).

A context menu item appears for Dual T1/E1/PRI, T1/E1/PRI, and Basic Rate modules that allows you to open the Busy Out Manager. You must be in the Controller view in Online Mode to access the Busy Out Manager.
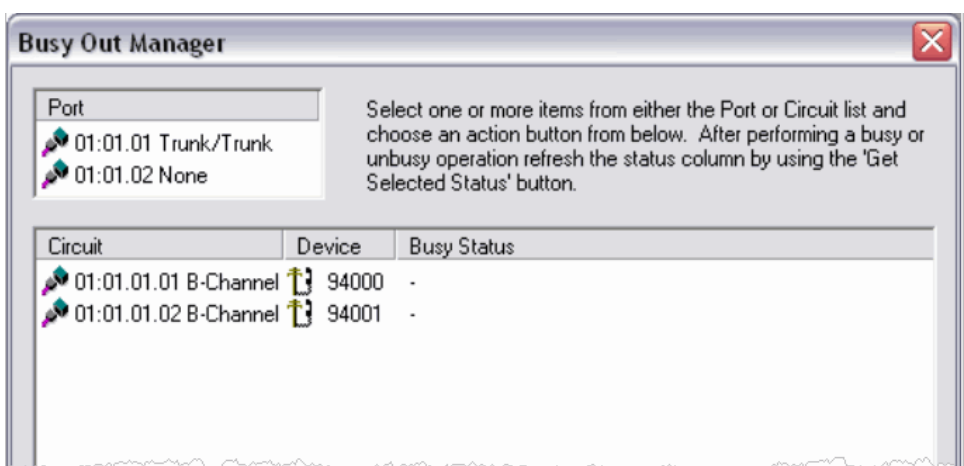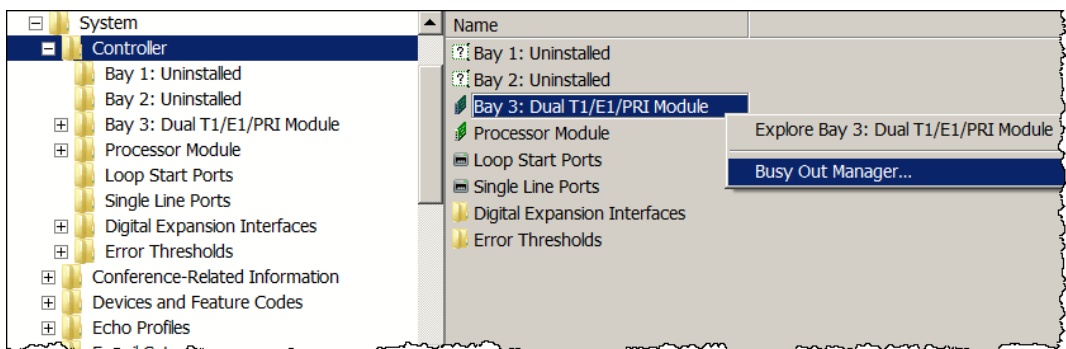
When you first open the Busy Out Manager for a particular module, the status of the devices/ circuits are unknown (shown as a '-') as they have not been retrieved yet. Anytime you want to see the "current" status of a particular item, you must select one or more items from either the ports view or devices/circuits view, and then click the **Get Selected Status**. If you choose to retrieve the status at the port level, the command may take a few seconds as it is retrieving status for every device/circuit programmed for the selected port(s).

You can similarly perform busy out operations on item selections from either the ports view or the device/circuits view.

To busy out items, do one of the following:

• Select one or more ports.

• Select one or more devices/circuits, and then click **Busy Selected**.

For example, if you want to busy out the entire board, select all of the ports in the port list, and then click **Busy Selected**.

Performing a "Busy Selected" action does not display the current status. The status for the devices/circuits instead displays "Pending Status Update," at which point you must retrieve the status.
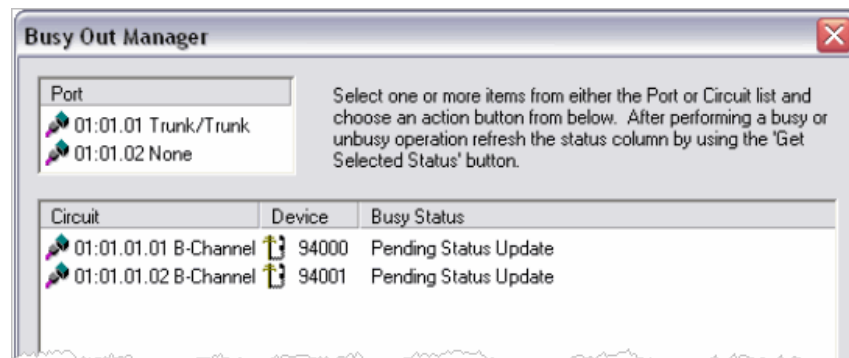
In the same way that you can perform busy out operations you can perform unbusy operations. Following the same steps as above, however, instead of clicking **Busy Selected** you would click **Unbusy Selected**.

Busy Out Statuses

There are several different statuses that may be shown in the Busy Out Manager. The display string and their meaning can be found in the table below.

Table 36:   Busy Out Statuses

STATUS	MEANING
-	The status of the device/circuit is unknown. To get the status, select the device/circuit, and then click **Get Selected Status**.
Not Scheduled For Busy Out	The device is operable. It may be idle or active, but it is not currently busied out or scheduled for busy out.
Busy	The device/circuit has been set to busy by the Busy Out Manager.
Pending Busy	The device has been scheduled to be busied out, but is currently in use.
Pending Status Update	The user has actioned an item for Busy/Unbusy. You need to retrieve the status.

POWER FAIL BYPASS CAPABILITIES

The HX controller has four loop start and four single line ports (the SL port addresses are 11.1.1, 11.2.1, 11.3.1, 11.4.1). The first two ports are used for the Power Fail Bypass feature. This feature allows up to two single line phones to be able to make calls after a total system shutdown due to power loss. The CS controller provides two loop start ports and two single line ports (the port addresses are 11.1.1 and 11.2.1). These ports are used for the Power Fail Bypass feature. Refer to the MiVoice Office 250 Installation Manual for information on how the ports are cabled.

PROGRAMMING ECHO THRESHOLDS

Specifications include several error thresholds. When an error exceeds one of these thresholds, an error message is sent to the error message output port (see **"Message Print"** on **page 77**). If necessary, you can set the thresholds for T1/E1/PRI errors in units of the number per hour and number per day. Thresholds are set to default values that, in most cases, will not need to be changed. Possible errors include the following:

- **BPV (Bipolar Violations)**: This is a non-zero signal element that has the same polarity as the previously received non-zero element.

- **CRC-6 (Cyclic Redundancy Check)**: The calculation carried out on a set of transmitted bits by the transmitter does not match the calculation performed by the system.

- **OOF/COFA (Out of Frame/Change of Frame Adjustment)**: OOF is a state in which the frame alignment that is received is not consistent with that which is transmitted. COFA occurs when the system realigns its receiver to the proper frame alignment signal. This could be caused by an incorrect line build-out (LBO) value.

- **CS (Controlled Slips)**: The system replicated or deleted one 192-bit digital signal (DS1) frame due to a lack of frequency synchronization. This could be caused by an incorrect line build-out (LBO) value.

- **ES (Errored Seconds)**: These are seconds in which at least one error occurred.

- **SES (Severely Errored Seconds)**: This is a second during which transmission performance is degraded below an acceptable level.

- **UAS (Unavailable Seconds)**: This is the time interval during which the T1, T1/PRI, or E1/PRI span is unavailable for service. This time begins with 10 or more consecutive Severely Errored Seconds and ends with 10 or more Non-Severely Errored Seconds.

To program a threshold:

1. Double-click **Error Thresholds** to view the list.

2. To change a specific threshold, click the Value or Daily field, and then enter the new number.

3. Press **ENTER** or click another field to save the change.

TROUBLESHOOTING MODULES

This section contains troubleshooting tips for the following issues:

- **Expansion Modules** below

- **T1/E1/PRI Modules** below

EXPANSION MODULES

Table 37 summarizes troubleshooting strategies recommended for resolving discrepancies involving expansion modules.

Table 37:   Expansion Module Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	POSSIBLE SOLUTION
The BRM-2 module ONLINE LED is off.	The module is not inserted all the way into the bay.	Make sure the module is inserted all the way into the bay and programmed correctly.
Cannot set the module to support station in NT mode.	The module does not support NT mode.	N/A

Table 37: Expansion Module Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	POSSIBLE SOLUTION
Cannot hook-up a video conference directly to a BRM-2 module.	BRM-2 does not support video conferences.	N/A
The module port is not synchronized to the CO [Local Exchange].	Reference clock may be incorrectly programmed.	Verify that the reference clock for the port is programmed correctly.

T1/E1/PRI MODULES

Table 38 summarizes the troubleshooting strategies recommended for resolving discrepancies that may occur with T1/E1/PRI (T1M) or Dual T1/E1/PRI (T1M-2) modules and trunks.

Table 38: T1/E1/PRI Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Dual T1/E1/PRI Module does not come online.	Hot-swap lever is not in the "on" position.	Make sure the module is properly installed in the bay. The lever is spring-loaded and should normally automatically move to the "on" position once the module is properly installed.
	Incorrect software.	Upgrade the MiVoice Office 250 software. The Dual T1/E1/PRI Module is supported in version 2.3 and later. Also make sure you have the latest Dual T1/E1/PRI package installed.
	Incorrect license.	Although the Dual T1/E1/PRI Module itself is not licensable, the second port is. Watch Message Print as you insert the module in the bay. If the board comes online, but the second port does not, you do not have a correct license. Otherwise, there is a different problem.
	Defective module.	If all of the above suggestions do not help, you may have a defective module. Please contact Mitel technical support.
A Dual T1/E1/PRI port does not come online.	Wrong license.	The second port on the T1M-2 has a separate license, named Additional T1/E1/PRI Ports. Regardless whether you first program port 1 or port 2, the second port you program requires a license. Make sure you have uploaded the correct license by looking at the License Information option in DB Programming.
One port or channel is always unavailable.	The port/channel has been busied out.	A channel, a port, or an entire module can be busied-out by a user. These can be unbusied either in DB Programming or in the DMU. You may also check the busyness status on the web page.

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Table 38:   T1/E1/PRI Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
T1 trunks have frequent "Controlled Slip" and "Out-Of-Frame" errors	Line Build-Out (LBO) setting incorrect	Use T1 Diagnostics to determine the current LBO status. Then use T1 Programming to change the setting. For more information, see **"Variable T1 Span Characteristics"** on **page 164.**
	Defective T1M or T1M-2 module	Check for inadequate connection. If the problem persists, contact the T1 provider or replace the module if faulty.

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TRUNK PROGRAMMING

Trunk programming includes the following:

- **Viewing or Programming Trunks** below
- **Changing Trunk Extension Numbers** below
- Copying Trunks on **page 220**
- Assigning Trunks to CO Trunk Groups on **page 220**
- Troubleshooting CO Trunks on **page 225**

VIEWING OR PROGRAMMING TRUNKS

You can view system trunks and program trunk options.

To view or program a specific trunk:

1. Select System – Devices and Feature Codes – **Trunks**.

NOTE	T1 or E1 trunks with a switch type of "Private Networking" or "IP Private Networking" appear in the trunk list, but you cannot change the options.

2. Double-click the trunk number. Trunk options appear in the right pane. To program trunk options, see **page 236**.

CHANGING TRUNK EXTENSION NUMBERS

You can change a single trunk number or use Batch Extension Exchange to change several trunk extension numbers at once.

To change a single trunk extension number:

1. Select System – Devices and Feature Codes – **Trunks**.

2. In the **Extension** column, select the trunk number, and then select the new number from the list.

3. Click out of the field or press **ENTER** to save the change.

To change several trunk extension numbers:

1. Select the trunks you want to change (you can use the SHIFT or CTRL key to select more than one trunk), right-click, and then select **Batch Extension Change**. The Create Extension dialog box appears.

2. Select the number that you want to assign to the first selected trunk (the other trunks will be numbered consecutively after this number).

3. Click **OK**. The trunks are automatically renumbered and resorted.

COPYING TRUNKS

To save time, you can copy a trunk and its settings instead of creating a new one.

To copy a trunk:

1. Select System – Devices and Feature Codes – **Trunks**.

2. Right-click the trunk extension, and then select **Copy**.

3. To paste the programming information into another trunk, right-click the trunk where you want the information pasted, then select **Paste**. The Copy *<trunk>* dialog box, similar to the one below, appears.
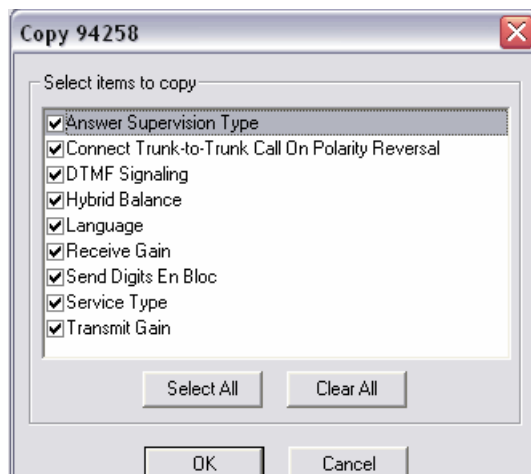
4. Select the attributes, and then click **OK**.

ASSIGNING TRUNKS TO CO TRUNK GROUPS

You can assign individual trunks to CO trunk groups. See **"CO Trunk Groups"** on **page 504** for more information about CO trunk groups.

To assign trunks to a CO trunk group:

1. Select System – Devices and Feature Codes – **Trunks**.

2. Double-click the trunk number.

3. Select **CO Trunk Group**.

4. Do one of the following:

Method A

a. In the **Value** column, select the current value, and then enter the new value in the box.

b. Click out of the field or press **ENTER**. The Select a Device dialog box appears.

c. Click **OK**. The new number appears in the field.

Method B

   a. Right-click the current CO Trunk Group value, and then select **Change CO Trunk Group**. A window appears prompting for the device type to include.

   b. Select **CO Trunk Group**, and then click **Next**. CO Trunk Groups with details appear in the right pane. To view them in a list only, click **List**.

   c. Select the trunk group, and then click **Finish**. Trunk groups with details appear in the right pane. You can also move trunks between trunk groups. See "Moving Trunks Between CO Trunk Groups" on **page 506**.

UNIFIED VOICE MESSAGING

Table 39 summarizes the troubleshooting strategies recommended for resolving discrepancies that may occur with Unified Voice Messaging (UVM). See "UVM Voice Processing System" on **page 838** for details.

Table 39:   Unified Voice Messaging Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE
UVM is running on the PS-1. IP phones or digital telephones have bad quality.	1. Voice packets are not getting from the Base Server to the PS-1 in a timely manner. This could happen for the following reasons: • Network is overloaded • PS-1 and Base Server are not on the same LAN and VLAN using the same switch 2. Voice packets are not getting from the PS-1 to the phones in a timely manner. This could happen if PS-1 or any of the IP phones have duplicate IP addresses. Power cycling can sometimes cause routers/switches to default to previous values.

CALLER ID FORWARDING

The following table lists troubleshooting information for the Caller ID Forwarding feature.

Table 40:   Caller ID Forwarding Support Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
The customer reports incoming trunk caller ID does not forward to an outbound PSTN call.	The original incoming trunk call did not have caller ID.	Enable service with the CO provider to ensure incoming trunk calls have caller ID.
	The system administrator has not enabled the "Propagate Original Caller ID for Unanswered Calls" flag for the correct CO trunk group.	Enable the "Propagate Original Caller ID for Unanswered Calls" flag for the correct CO trunk group.
	The outbound call is sent out a non-ISDN PSTN line.	For Caller ID Forwarding, the customer must use an ISDN line to forward caller ID.
	The original trunk is answered by the system (either through a Voice Mail application, human operator, or Unified Communicator).	Currently, the system does not support propagating caller ID for incoming trunk calls that are answered and then transferred back out to the PSTN.
	If the answer is none of the above, the CO may not support the ISDN message for propagating caller ID.	Call the CO provider to determine why the ISDN caller ID information is not forwarded. If the CO cannot resolve this issue, contact Technical Support and provide the following information: 1. the ISDN switch type where outbound calls was made; and 2. freeze of this call scenario.
The customer reports phone caller ID does not forward to external PBX.	The system administrator has not enabled the "Send Station Extension/Username to Attached PBX" flag for the correct CO trunk group.	Enable the "Send Station Extension/Username to Attached PBX" flag for the correct CO trunk group.
	The outbound call is sent out a non-ISDN PSTN line.	For Caller ID Forwarding, the customer must use an ISDN line to forward caller ID.
	An unsupported SIP gateway is used to connect the system to an external PBX.	Verify that the SIP gateway has been officially tested. If the SIP gateway is officially supported, call Technical Support and provide the following information: 1. the ISDN switch type where outbound calls was made, 2. type of gateway used to connect to external PBX, 3. ISDN switch type settings on SIP gateway, and 4. freeze of this call scenario.

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Table 40:   Caller ID Forwarding Support Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
The customer reports phone caller ID name (and only name) is not propagating to an external PBX. The customer is using a SIP gateway to communicate to an external PBX.	Two specific conditions cause this to happen: 1. the outbound ISDN line is configured as a DMS-100; and 2. the originating extension is calling out ARS and ARS is routing the call to a remote node.	There is a scenario where phone Caller ID Forwarding is used over a networked ARS configuration, and the phone user name will not be passed in the ISDN calling party name field. If the outgoing ISDN line uses the "display IE" method to transfer the calling party name, the calling party name will not use the phone's user name. Instead of the user name, the system uses the calling party name as the name.

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CALLER ID PROPAGATION

The following table lists troubleshooting information for the Caller ID Propagation feature. For complete information about Caller ID Propagation, see "Caller ID Propagation" on .

Table 41:   Caller ID Propagation Support Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Incoming trunk caller ID does not propagate to an outbound PSTN call.	The original incoming trunk call did not have caller ID.	Enable service with the Central Office (CO) provider to ensure incoming trunk calls have caller ID.
	The "Propagate Original Caller ID for Unanswered Calls" flag for the correct CO trunk group is not enabled.	Enable the "Propagate Original Caller ID for Unanswered Calls" flag for the correct CO trunk group.
	The outbound call is sent out on a non-ISDN PSTN line.	For Caller ID Propagation, the customer must use an ISDN line to propagate caller ID.
	The original trunk is answered by the system (either through a Voice Mail application, human operator, or Unified Communicator).	Currently, the system does not support propagating caller ID for incoming trunk calls that are answered and then transferred back out to the PSTN.
	If the cause is none of the above, the Central Office may not support the ISDN message for propagating caller ID.	Call the CO provider to determine why the ISDN caller ID information is not propagated. If the CO cannot resolve this issue, contact Mitel Technical Support and provide the following information: 1. the ISDN switch type where outbound calls were made; and 2. freeze of this call scenario.

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Table 41:   Caller ID Propagation Support Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Caller ID does not propagate to external PBX.	The "Send Station Extension/ Username to Attached PBX" flag is not enabled for the correct CO trunk group.	Enable the "Send Station Extension/ Username to Attached PBX" flag for the correct CO trunk group.
	The outbound call is sent out on a non-ISDN PSTN line.	For Caller ID Propagation, the customer must use an ISDN line to propagate caller ID.
	An unsupported SIP gateway is used to connect the system to an external PBX.	Verify that the SIP gateway has been officially tested. If the SIP gateway is officially supported, call Mitel Technical Support and provide the following information: 1. the ISDN switch type where outbound calls were made; 2. the type of gateway used to connect to external PBX; 3. ISDN switch type settings on SIP gateway; and, 4. freeze of this call scenario.
The customer reports station caller ID name (and only name) is not propagating to an external PBX. The customer is using a SIP gateway to communicate to an external PBX.	Two specific conditions cause this to happen: 1. the outbound ISDN line is configured as a DMS-100; and 2. the originating extension is calling out ARS and ARS is routing the call to a remote node.	There is a scenario where station Caller ID Propagation is used over a networked ARS configuration, and the station username will not be passed in the ISDN calling party name field. If the outgoing ISDN line uses the "display IE" method to transfer the calling party name, the calling party name will not use the station's username. Instead of the username, the system uses the calling party name as the name.

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TROUBLESHOOTING CO TRUNKS

Table 42 summarizes the troubleshooting strategies recommended for resolving CO [Local Exchange] trunk discrepancies.

Table 42: CO Trunk Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
CO trunk inoperative throughout the system	Defective CO trunk from central office	At the associated CO block, remove the bridging clips for the trunk. On the telco side of the block, use a test set to verify the CO trunk connection. Also, move the CO trunk to a known good CO circuit. If the problem follows the trunk, contact the telephone company.
	Trunk is dedicated to a secondary carrier requiring an access code	Verify the type of CO trunk. Instruct users to dial access code if required.
	Defective cabling or mis-wired amphenol connector on the trunk module	Using a test set, ensure presence and correct location of the CO trunk at the associated CO block.
	Programming error	Ensure that the trunk is assigned to the correct CO trunk group and phones have been given access to it. See "Assigning Trunks to CO Trunk Groups" on **page 220**.
	Defective trunk module	Replace the associated trunk module.
	Defective processor module	Replace the module if faulty.
Cannot obtain CO dial tone	If reorder tone is heard, programming error	Ensure that phone has outgoing access. Also, check to see if the phone is programmed for ARS only. Ensure trunk is assigned to the correct trunk group. See "Assigning Trunks to CO Trunk Groups" on **page 220**.
	If progress tone is heard, user error	System is programmed to expect a forced account code. See "Phone/Phantom/Hot Desk Profile Flags" on **page 417**.
	Selected trunk is designated as incoming-only	Cannot dial out on an incoming-only trunk.
	Defective trunk module	Replace the associated trunk module.
	Defective processor module	Replace the module if faulty.

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Table 42:   CO Trunk Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Low volume on all CO trunks; cannot break CO dial tone	Defective CO trunk from central office	At the CO block, remove the bridging clips for the trunk. On the telco side of the block, use a test set to verify the CO trunk connection. Also, move the CO trunk to a known good CO circuit. If the problem follows the trunk, contact the telephone company.
	Open or loose connection in the cable between the power supply and the backplane	If, with all trunk modules removed, the system volt ages are still not within tolerance, turn off the system and check the cable and connector with an ohmmeter. Replace or repair the faulty cable.
	Defective power supply	Replace the chassis if the power supply is faulty.
Cannot break CO dial tone	CO circuit is programmed for wrong signaling type	Ensure that CO trunk and CO circuit use same type signaling (DTMF or dial pulse). See "DTMF Signaling" on **page 238**.
	Defective CO trunk from central office	At the CO block, remove the bridging clips for the trunk. On the telco side of the block, use a test set to verify the CO trunk connection. Also, move the CO trunk to a known good CO circuit. If the problem follows the trunk, contact the telephone company.
	Timer error	If trunk is DTMF, the DTMF Digit Duration/ Pause timer setting may not be compatible with the trunk. See "Timers and Limits" on **page 756**.
	Defective trunk module	Replace the associated trunk module.
	Defective processor module	Replace the module if faulty.

Note: See also phone problems on "Troubleshooting Phones" on **page 483**.

Table 42:   CO Trunk Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Cannot place an outgoing call; CO dial tone is present; intercom works	Programming error	Check phone class of service (COS). Check that equal access and absorbed digit programming for the trunk group are correct. Check ARS. See **"Toll Restrictions"** on **page 511**.
	Defective CO trunk from central office	At the CO block, remove the bridging clips for the trunk. On the telco side of the block, use a test set to verify the CO trunk connection. Also, move the CO trunk to a known good CO circuit. If the problem follows the trunk, contact the telephone company.
	Defective phone	Replace the phone and/or perform the phone self-test as outlined in the Installation chapter in the *MiVoice Office 250 Installation and Administration Manual* .
	Selected trunk is designated as incoming-only	Cannot dial out on an incoming-only trunk.
	Defective trunk module	Replace the associated trunk module.
	Defective processor module	Replace the module if faulty.

Note: See also phone problems on "Troubleshooting Phones" on **page 483**.

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Other phone conversations can be heard on the CO trunk (crosstalk)	Defective CO trunk(s)	Isolate the trunk(s) with crosstalk by removing the bridging clips from the CO block. On the telco side of the block, attach a test set to each trunk and check for crosstalk. If present, contact the telephone company.
	Defective cabling or mis-wired amphenol connector on the trunk module	Using a test set, ensure presence and correct location of the CO trunk at the associated CO block.
	Defective digital telephone module	Replace the associated Digital Endpoint Module.
	Defective trunk module	Replace the associated trunk module.
	Defective processor module	Replace the module if faulty.

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Table 42:   CO Trunk Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Outside calls dropped during conversation or upon answering CO trunk	User error (trunk button being pressed after initial connection is made)	Instruct users not to press the trunk button while on a call. Or, if necessary, set the CO Reseize timer to a higher value. The default value is 3 seconds. Or, program the phone to disable CO reseize. See "Timers and Limits" on **page 756**.
	Defective CO trunk from central office	At the CO block, remove the bridging clips for the trunk. On the telco side of the block, use a test set to verify the CO trunk connection. Also, move the CO trunk to a known good CO circuit. If the problem follows the trunk, contact the telephone company.
	Insufficient loop current supplied by central office	Central office must supply 18 mA minimum loop current.
	IC-CO Disconnect timer value is set too short	Ensure timer value is long enough to ignore normal interruptions in CO loop current. The default value is 0.6 seconds. See "Timers and Limits" on **page 756**.
	Defective Digital Endpoint Module	Replace the phone and/or perform the phone self-test as outlined in Installation chapter in the *MiVoice Office 250 Installation and Administration Manual* .
	Defective trunk module	Replace the trunk module if faulty.
	Defective processor module	Replace the module if faulty.
CO trunk cannot be reseized	CO Reseize feature is disabled	Check phone programming. If the CO Reseize option has been disabled, the user cannot reseize a trunk until it has been disconnected by hanging up or pressing another trunk button. See "Timers and Limits" on **page 756**.
	CO Reseize timer is set too high	Set the timer to a lower timer value. The default value is 3 seconds. See "Timers and Limits" on **page 756**.
	User error	If the phone is programmed with the validate account codes option and the user enters an invalid Account Code For All Calls Following, calls will not be allowed at the phone until the code is removed or reprogrammed. See "Phone/Phantom/Hot Desk Profile Flags" on **page 417**.

Table 42: CO Trunk Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Noise on CO trunk at all phones	Defective CO trunk	At the CO block, remove the bridging clips for the trunk. On the telco side of the block, use a test set to verify the CO trunk connection. Also, move the CO trunk to a known good CO circuit. If the problem follows the trunk, contact the telephone company.
	Defective trunk module	Replace the associated trunk module.
	Defective power supply	Replace the chassis if the power supply is faulty.
	Defective processor module	Replace the module if faulty.
CO trunk remains seized after a call has been ended	Characteristic of some ESS central offices	Central office must provide disconnect signal. Or, install a trunk release module available from most supply houses.
	IC-CO or CO-CO Disconnect timer value set too long	Central office disconnect signal was not detected by IC-CO or CO-CO Disconnect timer. The default value of the IC-CO timer is 0.6 seconds; the CO-CO timer is 0.35 seconds. See "Timers and Limits" on **page 756**.
	Defective CO trunk	At the CO block, remove the bridging clips for the trunk. On the telco side of the block, use a test set to verify the CO trunk connection. Also, move the CO trunk to a known good CO circuit. If the problem follows the trunk, contact the telephone company.
	Defective phone	Replace the phone and/or perform the phone self-test as outlined in the Installation chapter in the *MiVoice Office 250 Installation and Administration Manual* .
	Defective trunk module	Replace the associated trunk module.
	Defective processor module	Replace the module if faulty.
The module port is not synchronized to the CO [Local Exchange].	Reference clock may be incorrectly programmed.	Verify that the reference clock for the port is programmed correctly.

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SIP GATEWAYS

> **NOTICE**
>
> SIP trunks and gateways require special configuration settings for emergency calls. For more information, see "Emergency Extensions for IP Devices" on **page 642**.

The system supports SIP trunks to connect to the Central Office (the public telephone network). SIP trunks allow the system to communicate with the CO through SIP-enabled gateways. If you are using SIP trunks, at least one SIP gateway must exist at all times, even in a default system, because it follows the model of a CO trunk group.

For a more current list of compatible SIP gateways, refer to KB article 09-4940-00056 on the Mitel Knowledge Base Center (http://domino1.mitel.com/prodsupp/prodsupkb.nsf/WebSearchForm?OpenForm). Note that you must first log on to Mitel OnLine (http://portal.mitel.com/wps/myportal/MOLHome) in order to access the KB.

Using a SIP Gateway is an older method whereby a hardware device is installed at a location to allow the MiVoice Office 250 to connect to trunks that are originally TDM, but are being converted to SIP by the gateway. To configure SIP Trunks to connect to a SIP Trunk Provider, see "Service Provider SIP Trunks and SIP Trunk Groups" on **page 691** for more information.

To view SIP Gateways and options:

Select System – Trunk-Related Information – **SIP Gateways**.

To view SIP Trunks:

Select System – Trunk-Related Information – SIP Gateways – *<gateway>* – **SIP Trunks**.

UNDERSTANDING NAT CHALLENGES FOR SIP DEVICES

A Network Address Translation (NAT) device translates private network IP addresses to one or more public network IP addresses based on NAT translation rules.

NAT devices are installed at the edge of the private network and have internal and external interfaces (and IP addresses). For outgoing IP traffic from the private network to the Internet, NAT translates the source IP address. For incoming IP traffic from the Internet to the private network, NAT translates the destination IP address.

NAT devices provide the following advantages:

- Internal IP addresses are hidden from the open Internet and therefore more secure.

- IP addresses are conserved because they are allocated dynamically when needed.

Despite the advantages of NAT devices, they can cause problems for protocols using Peer-to-Peer technologies like multimedia traffic on VoIP networks using SIP.

PLACING A SIP GATEWAY BEHIND A NAT DEVICE

Applies only to IP gateway trunks for SIP gateways. If you are using SIP trunks, at least one SIP gateway must exist at all times, even in a default system, because it follows the model of a CO trunk group.

You can place a SIP gateway behind a NAT device. For more information about NAT devices, see "Understanding NAT Challenges for SIP Devices" on **page 230** and refer to "Appendix B: Network IP Topology," in the *MiVoice Office 250 Installation and Administration Manual* .

To place a SIP gateway behind a NAT device, you must program the following two fields:

- **System NAT IP Address**: To establish the NAT, or public, IP address.

- **SIP Gateway Name**: To inform the gateway where the SIP messages are originating.

To program the System NAT IP Address:

1. Select System **–** IP Settings – **System NAT IP Address**.

2. In the **Value** column, type the NAT, or public, IP address that the system has on the NAT side of the firewall. For a description of the field, see **page 608**.

To program SIP Gateway options:

1. **Select System** – Trunk-Related Information – SIP Gateways – <*gateway*>.

2. Enter the gateway that is programmed on the SIP gateway itself. The default value is DEFAULT. If you attempt to set the SIP Gateway Name to an empty string, the system displays the following error message, and denies the change, as shown in the following illustration.
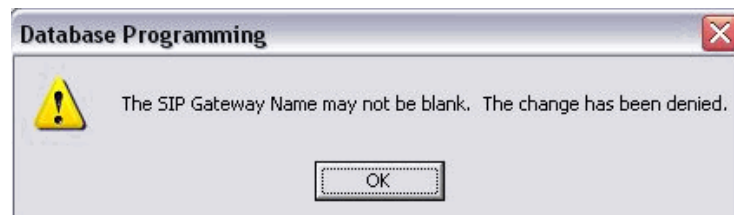
3. Program the following options:
    - **SIP Trunks**: The SIP gateway where SIP trunks are placed.
    - **IP Address**: The IP address of the SIP gateway.
    - **Listening Port**: The port that the system "listens" to for SIP messages. By default, this port is set to 5060 (standard SIP port).

4. Click out the fields or press **ENTER** to save the changes.

GATEWAY SIP TRUNKS

Gateway SIP trunks allow the system to communicate with the CO through SIP gateways. For more information about SIP gateways, see "SIP Gateways" on **page 230**.

SIP trunks:

- Are transparent to the system user because SIP trunks function like any other CO trunk in the system.

- Support transferring trunks, putting trunks on hold, and connecting trunks to conferences similar to other CO trunks in the system.

- Support making and receiving calls by any phone.

- Support Peer-to-Peer Media by IP phones.

- Reside in CO trunk groups just like other trunks so that SIP trunk calls can be routed using Automatic Route Selection (ARS).

CREATING SIP TRUNKS

You can also use the Configuration Wizard to create SIP trunks. See "Launching the Configuration Wizard" on **page 59** for instructions on how to launch the wizard. For complete information about the Configuration Wizard, refer to the *MiVoice Office 250 DB Programming Help*. The following procedure describes creating SIP trunks through a SIP gateway. See "Service Provider SIP Trunks and SIP Trunk Groups" on **page 691** for information on creating SIP Peer trunks.

To create a SIP trunk:

1. Select System – Devices and Feature Codes – **Trunks**.

2. Right-click in a blank area in the right pane, and then select **Create SIP Trunk**. The Create SIP Extension dialog box appears.

3. Enter the starting extension number and the number of extensions, and then click **OK**. The SIP trunk appears in the list.

PROGRAMMING SIP TRUNK OPTIONS

To program SIP trunk options:

1. Select System – Devices and Feature Codes – **Trunks**.

2. Double-click the SIP trunk. SIP trunk options are shown in the right pane.

3. Program the following options:

    - Call Configuration on **page 244**
    - CO Trunk Groups on **page 504**
    - Network Group on **page 243**
    - SIP Gateways on **page 230**
    - CP History on **page 245**

- Language on **page 242**

MGCP GATEWAYS, DEVICES, AND TRUNKS

Beginning with release 6.0 SP1, MGCP gateways, devices, and trunks are no longer supported. These have been replaced with SIP gateways and more recently, SIP Peer Trunks.

> ### NOTICE
>
> Media Gateway Control Protocol (MGCP) trunks and gateways require special configuration settings for emergency calls. For more information, see "Emergency Extensions for IP Devices" on **page 642**.

The following sections provide instructions to add and configure MGCP devices. You can also use the Configuration Wizard to add and configure MGCP devices. See "Launching the Configuration Wizard" on **page 59** for instructions on how to launch the wizard. For complete information about the Configuration Wizard, refer to the *MiVoice Office 250 DB Programming Help*.

CREATING AN MGCP GATEWAY AND PHONE

Because you must assign MGCP phones to MGCP gateways when you create new MGCP phones, you must create an "MGCP Gateway and Phone" before you can add an MGCP phone. See "Adding an MGCP Phone" on **page 235**.

To create an MGCP gateway and phone:

1. Under System – Devices and Feature Codes – **Trunks**, right-click in the right pane.

2. Select **Create MGCP Gateway and Phone**. The following dialog box appears.
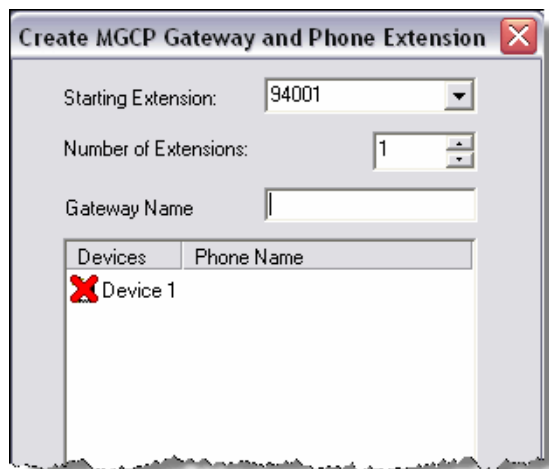
3. In the **Starting Extension** list, either use the default starting extension shown or select a different extension from the list.

4. In the **Number of Extensions** box, enter the number of extensions either by clicking the up and down arrows or by typing the number in the box.

5. In the **Gateway Name** box, type the gateway name.

6. Click any device with a red "X," and then type a name.

7. Click **OK**.

8. Configure trunk options as necessary. See "Trunk Programming Options" on **page 236**.

ADDING AN MGCP PHONE

To create MGCP phones:

1. Under System – Devices and Feature Codes – **Trunks**, right-click in the right pane.

2. Select **Create MGCP Phone**. The Create MGCP Phone Extension dialog box appears.

3. In the **Starting Extension** list, either use the default starting extension shown or select a different extension from the list.

4. In the **Number of Extensions** box, enter the number of extensions either by clicking the up and down arrows or by typing the number in the box.

5. In the **Associated Gateway** list, select the MGCP gateway to be assigned to the phones.

6. Click any device with a red "X," and then type a name.

7. Click **OK**.

8. Configure trunk options as necessary.

CHANGING THE MGCP GATEWAY IP ADDRESS

Applies only to IP gateway trunks for MGCP gateways. The MGCP Gateway IP Address option identifies the IP address used to access the MGCP gateway.

To enter an IP address:

1. Click the current Value. The Edit MGCP Gateway IP Address dialog box appears.

2. Without including the periods, enter the IP address, and then click **OK**. The default IP address is 192.168.200.201 (entered as 192168200201).

TRUNK PROGRAMMING OPTIONS

You can program the following trunk options. The options described in this section vary depending on trunk type. See **Table 43** for available options for each trunk type.

To program trunk options:

Select System – Devices and Feature Codes – Trunks – *<trunk>*.

Table 43:   Trunk Types and Feature Options

TRUNK OPTION	REF. PAGE	ANALOG LS (LSM)	DIGITAL LS (T1)	GS (T1)	DID (T1)	B-CHANNEL (T1/PRIOR E1/PRI)	E&M (T1)	MGCP GTWY. AND PHONE	MGCP PHONE	SIP
Answer Supervision Type	239	✓	✓	✓			✓	✓	✓	
Associated Gateway	246								✓	
Call Configuration	244							✓	✓	✓
Trunk Group	237	✓	✓	✓	✓	✓	✓	✓	✓	✓
Communication Timeout	245							✓		
Connect Trunk-to-Trunk Call On Polarity Reversal	240	✓				✓		✓	✓	
Connected to CO	241	✓								
CP History	245									✓
Disconnect Timer	239				✓		✓			
DTMF Signaling	238	✓	✓	✓	✓		✓	✓	✓	
Echo Profile	241	✓	✓	✓	✓	✓	✓	✓	✓	✓
Phone Name	246							✓	✓	
Gateway Name	246							✓		
Hybrid Balance	241	✓						✓	✓	
Language	242	✓	✓	✓	✓	✓	✓	✓	✓	✓
Last Hybrid Balance Test Timestamp	241	✓								
Manufacturer	246							✓		
Effective Echo Return Loss (ERL)	242	✓								
MGCP Gateway IP Address	235							✓		
MGCP Gateway Port	245							✓		

Table 43:   Trunk Types and Feature Options (continued)

TRUNK OPTION	REF. PAGE	ANALOG LS (LSM)	DIGITAL LS (T1)	GS (T1)	DID (T1)	B-CHANNEL (T1/PRIOR E1/PRI)	E&M (T1)	MGCP GTWY. AND PHONE	MGCP PHONE	SIP
NAT Address Type	**244**							✓	✓	✓
Network Group	**243**							✓	✓	✓
Number of Digits to Receive	**242**				✓		✓			
Reserve IP Resource for Device	**243**							✓	✓	✓
Send Digits En Bloc	**241**	✓	✓					✓	✓	
Service Prefix Base Number	**242**				✓	✓	✓			
Service Type	**237**	✓	✓				✓	✓	✓	
SIP Gateway	**231**									✓
Span Echo Profile	**735**	✓[1]	✓[1]	✓[1]	✓	✓*				
Start Type	**238**				✓		✓			

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1.   Available for Dual T1/E1/PRI modules only

CO TRUNK GROUP

Program CO trunk group options. For more information, see "Programming Trunk Group Options" on **page 506**.

SERVICE TYPE

You must program each trunk, not the trunk group, to collect digits using Caller ID [CLID], DID [DDI], ANI, DNIS, or DNIS-ANI. If the trunk is programmed to collect digits, but the trunk group does not use call routing tables, the system routes the call based on the trunk group programming and ignores the collected digits. However, the collected digits do appear in SMDR and can be used by the Desktop Interface. If the trunk is not set up to collect digits and the trunk group uses call routing tables, the system uses the call routing destination for "no-digit" calls.

Desktop Interface functionality requires the Desktop Interface software license, part no. 840.0319.

Available service types are determined by the trunk type. Service types are as follows:

- **Loop Start** and **MGCP Gateway and Phone**: *Do not change the Service Type to Caller ID unless you are using IP SLAs.* Loop start trunks can use Caller ID [CLID]. When selected, the Caller ID [CLID] service type option indicates that the associated trunk provides caller identification signals. Because the CO [local exchange] sends the information after the first cycle of ring voltage is applied, selecting this box also prevents the system from signaling an incoming call to the phone users until the system has had an opportunity to collect the

caller information. If no caller information is collected, the system provides ring signaling on the second ring signal. Caller ID [CLID] uses Caller ID [CLID] receivers. The following Caller ID standards are available for Loop Start trunks:

- Bellcore FSK Caller ID

- British Telecom FSK Caller ID

NOTE	For Release 6.1 and higher, MiVoice Office 250 allows the decoding of Calling Line Identification (CLID) information in countries which use variations of BT CLIP (British Telecom Calling Line Identification Presentation) signaling on analog trunks. BT CLIP supports the delivery of Caller ID information on analog trunks in the UK, and the standard is commonly known as BT CLIP. BT CLIP is an on-hook capability that provides the user with information about the caller before actually answering a call. BT CLIP replaces ETSI DTMF Caller ID.

In Europe, the loop start trunk provides caller identification signals when used with an MGCP gateway.

- **E&M and DID [DDI]**: E&M and DID trunks can use DID, E&M, ANI, or DNIS.

- **Ground Start and B-Channel**: Ground start and B-channel trunks do ***not*** support any of the caller information service types.

To set the Service Type:

1. Select System – Devices and Feature Codes – Trunks – <***trunk number***>.

2. Select **Service Type**.

3. In the value column, select the service type from the list.

4. Click outside the field or press **ENTER** to save the change.

DTMF SIGNALING

Does not apply to SIP and B-channel trunks. Other trunks use DTMF signals. Do not disable this flag. It is enabled by default.

NOTE	Enable this option for MGCP trunks; otherwise, the system may receive double digits.

START TYPE

Applies only to E&M and DID [DDI] trunks only. E&M and DID trunks perform a "handshake" between the system and the CO [local exchange] to transfer call information. The following are E&M and DID Start Types:

- **Immediate**: The calling system immediately begins sending the dialed digits to the receiving system.

- **Wink**: The systems perform a "handshake" to allow the receiving system to signal that it is ready to receive the digits dialed by the other system. This is the default start type.

- **Delay Dial**: The calling system waits until its E&M Dial Delay timer expires before sending any digits to the receiving system.

- **Dial Tone** *(E&M trunks only):* The calling system waits until it detects dial tone from the other system before sending digits.

To set the Start Type:

1. Select System – Devices and Feature Codes – Trunks – <***E&M or DID trunk***>.

2. Select **Start Type**.

3. In the **Value** column, select the option from the list.

4. Click out of the field or press **ENTER** to save the change.

DID DISCONNECT TIMER

Applies only to E&M and DID [DDI] trunks only. You can program trunks to use the Disconnect timer to disconnect calls immediately whenever the inside party hangs up on a call. If the Disconnect timer is not enabled, it waits up to two seconds for an on-hook signal from the CO [local exchange].

To set the Disconnect Timer option:

1. Select System – Devices and Feature Codes – Trunks – <***trunk number***>.

2. Select **E&M and DID Disconnect Timer**.

3. In the **Value** column, select either **Disconnect Timer** or **Wait for Remote On-Hook**.

4. Click out of the field or press **ENTER** to save the change.

ANSWER SUPERVISION TYPES

Applies to loop start, ground start, and MGCP gateway and phone trunks only. The Answer Supervision Type determines whether the system should consider a call valid or disconnected. Available Answer Supervision Types depend on whether you are programming a loop-start or E&M trunk:

For Loop Start, Ground Start, or MGCP Gateway and Phone trunks:

- **Polarity Reversal**: A loop reversal must be received to consider the call valid. When the first loop reversal is received, the call is made valid immediately, and the phone display begins call cost. When a second loop reversal is received, the system terminates the call. If a second loop reversal is not received, the system does not terminate the call unless the inside party hangs up or loss-of-loop is received from the CO [local exchange].

- **Valid Call Timer**: After the Valid Call Timer expires, the call is validated. All polarity reversals received before and after the Valid Call Timer are ignored.

- **Valid Call Timer with Polarity Reversal**: If a loop reversal is received before the Valid Call Timer expires, the call is made valid immediately, and the phone display begins call cost. When a second loop reversal is received, the system terminates the call. If a loop reversal is not received before the Valid Call Timer expires, the call is validated by the timer. If a loop reversal is received after the timer expires, the loop reversal is ignored, but the call cost is reset. If a second loop reversal is then received, the system terminates the call. If a second loop reversal is not received, the system does not terminate the call unless the inside party hangs up or loss-of-loop is received from the CO [local exchange].

For E&M trunks:

- **Use Off-Hook Debounce Timer**: After the Use Off-Hook Debounce Timer expires, the call is validated. All polarity reversals received before and after the Off-Hook Debounce Timer duration are ignored.

- **Use Answer Recognition Timer**: After the Use Answer Recognition Timer expires, the call is validated. All polarity reversals received before and after the Answer Recognition Timer duration are ignored.

To set the Answer Supervision Type:

1. Select System – Devices and Feature Codes – Trunks – <***Loop Start, Ground Start, or MGCP Gateway and Phone trunk***>.

2. Select **Answer Supervision Type**.

3. In the **Value** column, select the option from the list.

4. Click out of the field or press **ENTER** to save the change.

CONNECT TRUNK-TO-TRUNK CALLS ON POLARITY REVERSAL

Applies to loop start, B-channel, MGCP Gateway and Phone, and MGCP Phone trunks only.

NOTE	For this feature to work properly, both the incoming (B-channel trunk) and outgoing (non-T1 Loop Start trunk) sides of the trunk-to-trunk call must have this flag enabled (Yes). For non T-1 Loop Start trunks, this flag is available only when the answer supervision is set to "Polarity Reversal" or "Valid Call Timer with Polarity Reversal."

Trunk-to-Trunk Calls on Polarity Reversal settings are as follows:

- **Enabled (Yes)**: When an ISDN incoming call is made to a loop start trunk (trunk-to-trunk call), the call is validated when a polarity reversal is received. The MiVoice Office 250 connects the voice path to hear progress tones and then sends the ISDN connect message when the loop start polarity reversal is detected.

- **Disabled (No)**: When an ISDN incoming call is made to a loop start trunk (trunk-to-trunk call), the call is validated after receiving an end of dialing handshake. After the handshake, the CO [local exchange] may provide progress tones, such as ringback.

To enable or disable Trunk-to-Trunk Calls on Polarity Reversal:

1. Select System – Devices and Feature Codes – Trunks – <***trunk number***>.

2. Select **Connect Trunk-to-Trunk Calls on Polarity Reversal**.

3. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

4. Click out of the field or press **ENTER** to save the change.

SEND DIGITS EN BLOC

Applies to loop start trunks and MGCP Gateway and phone trunks only. The Send Digits En Bloc option determines how the system sends digits to the trunk. If enabled (set to Yes), the system sends all of the available digits when the system seizes the trunk (for example, when the user accesses ARS). If disabled (set to No), the system sends the digits to the trunk one at a time. By default, this is disabled, which is recommended for loop start trunks and MGCP gateways/phones.

To enable or disable Sending Digits En Bloc:

1. Select System – Devices and Feature Codes – Trunks – <***Loop Start, MGCP Gateway and Phone, or MGCP Phone trunk***>.

2. Select **Send Digits En Bloc**.

3. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

4. Click out of the field or press **ENTER** to save the change.

ECHO PROFILE

All system devices are associated with an echo profile. For more information and programming instructions, see "Echo Profiles" on **page 735**.

CONNECTED TO CO

Indicates whether or not the trunk is connected to another PBX or the CO. The default setting is Yes, because most systems connect trunks to the CO.

To enable the Connected to CO option:

1. Select System – Devices and Feature Codes – Trunks – <***trunk number***>.

2. Select **Connected to CO**.

3. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

4. Click out of the field or press **ENTER** to save the change.

HYBRID BALANCE

Applies to analog loop start trunks only. The (improved) Hybrid Balance Test automatically measures and determines the best hybrid balance setting for each type of trunk.

You can start an Automatic Hybrid Balance Test on a single trunk or all trunks. For more information, see "Hybrid Balance Test" on **page 1121**.

MEASURED ECHO RETURN LOSS

This read-only value is the Finalized Hybrid Balance Test – ERL measured in the most recent Hybrid Balance Test on an analog loop start trunk. This value displays in Message Print output. A red "X" appears on digital loop start trunks. For more information, see "Hybrid Balance Test" on **page 1121**.

SERVICE PREFIX BASE NUMBER

Applies to DID [DDI] and E&M trunks that are programmed for the DID (DDI for Europe), ANI, DNIS, *DNIS*, and *ANI*DNIS* service types only. This is the base number for the trunk. If you are using B-channel trunks, the CO [local exchange] sends fewer than 10 digits for the DNIS number on some ISDN calls. When this happens, the system must construct a 10-digit number for the System Open Architecture Interface (OAI) and Desktop Interface features (both of these features require a software license). To construct the number, the system requires a base number for the trunk.

To set the Service Prefix Base Number:

1. Select System – Devices and Feature Codes – Trunks – <**DID or E&M trunk**>.

2. Select **Service Prefix Base Number**.

3. In the **Value** column, select the current value, and then enter the digits in the box (up to 48 digits).

4. Click out of the field or press **ENTER** to save the change.

NUMBER OF DIGITS TO RECEIVE

Applies to DID and E&M trunks that are programmed for the DID [DDI], ANI, and DNIS service types only. This is the number of digits the system uses for directing calls.

To set the Number of Digits to Receive:

1. Select System – Devices and Feature Codes – Trunks – <**trunk number**>.

2. Select **Number of Digits to Receive**.

3. In the **Value** column, type or select the correct amount (1–32) in the box.

4. Click out of the field or press **ENTER** to save the change.

LANGUAGE

You can select the language used for voice prompts and displays when this trunk is used. Options include Use Primary Language, Use Secondary Language, American English, British English, Canadian French, Japanese, or Spanish. See "Multilingual Capability" on **page 314** for details about supported languages.

NOTE The Japanese language is not supported on 52xx\53xx phones.

To set the language:

1. Select System – Devices and Feature Codes – Trunks – <***trunk number***>.

2. Select **Language**.

3. In the **Value** column, select the language from the list.

4. Click out of the field or press **ENTER** to save the change.

NETWORK GROUP

Applies to IP gateway trunks for MGCP and SIP gateways and MGCP phones only. The Network Group option is required only if you are using peer-to-peer (P2P) media (audio or video) for the selected MGCP gateway, MGCP phone, or SIP gateway. This option is also located under System – Devices and Feature Codes – **Network Groups**. See "Network Groups" on **page 588**.

To assign the gateway or phone to a network group:

1. Select System – Devices and Feature Codes – Trunks – <***trunk number***>.

2. Double-click **Network Group**.

3. Double-click the desired Network Group extension, and then add the gateway or phone to the IP Trunks list.

4. Click out of the field or press **ENTER** to save the change.

RESERVE IP RESOURCES FOR DEVICE

You can program IP trunks (and IP phones) with a dedicated IP resource. When the Reserve IP Resource for Device field is enabled (set to Yes), an IP Resource is reserved for the device. You can also use the Resource Reservation Tool to program this option. See "Resource Reservation Tool" on **page 647**.

| NOTE | Mitel recommends that you reserve IP resources for attendants and other high traffic users (for example, call center agents). However, excessive use of reservations degrades the effectiveness of oversubscription by reducing the amount of resources available to be shared. |

If a device that has a reserved IP resource goes off-line in Call Processing, the IP resource returns to the shared pool of available IP resources. Although Call Processing makes this adjustment, the database does not change until you delete the device or change this flag.

To reserve or unreserve an IP resource for a particular IP trunk:

1. Select System – Devices and Feature Codes – Trunks – <***trunk number***>.

2. Select the type of IP trunk device, MGCP gateway or SIP trunk.

3. Select **Yes** to reserve a resource for the device, or select **No** to unreserve the resource. The default setting is No.

4. Click out of the field or press **ENTER** to save the change.

NAT ADDRESS TYPE

Applies to MGCP Gateway and Phone or SIP Gateway trunks only. This option specifies whether to use the Native or NAT IP Address for audio commands. For more information, see "Understanding NAT Challenges for SIP Devices" on **page 230**, and refer to Appendix B: Network IP Topology, in the *MiVoice Office 250 Installation and Administration Manual*

To set the NAT Address Type:

1. Select System – Devices and Feature Codes – Trunks – <***trunk number***>.

2. Double-click the trunk extension or CO [Local Exchange] trunk group number.

3. Click **NAT Address Type**, and then select either **Native** or **NAT**. The default setting is Native.

4. Click out of the field or press **ENTER** to save the change.

CALL CONFIGURATION

Applies only to IP gateway trunks for MGCP gateways and phones and for SIP gateways. The Call Configuration information is only required if using P2P media for the selected MGCP gateway, phone, or SIP gateway. The Call Configuration defines the settings that the gateways and phones use when connected to a call. For more information about Call Configurations, see "IP Call Configurations" on **page 629**.

To assign the IP device to a call configuration:

1. Select System – Devices and Feature Codes – Trunks – <***trunk number***>. (You can also configure this option from System – IP-Related Information – **Call Configurations**.)

2. Double-click **Call Configuration**.

3. Double-click the Call Configuration ID.

4. Double-click **Trunks**.

5. Right-click in the right pane, and then click **Move to Trunks list**. The Move to Trunks List dialog box appears.

6. Select the gateway or phone to the IP Trunks list, and then click **Next**.

7. Click **Move Items**, and then click **Finish** to save the change.

To change the call configuration:

1. Right-click the trunk number, and then select **Change Call Configuration**. The Change Call Configuration dialog box appears.

2. Select **Call Configuration**, and then click **Next**. The devices with details appear. To view items in a list only, click **List**.

3. Click **Finish**.

CP HISTORY

Applies to SIP trunks only. The CP History is used for diagnostic purposes only. It enables SIP message output in the Call Processing history file. You can use the CP History to trace SIP trunk calls. The system-wide SIP format is set to No Output by default.

To change the CP History setting:

1.   Select System – Devices and Feature Codes – Trunks – <***SIP trunk***>.

2.   Select **CP History**.

3.   In the **Value** column, select the option from the list. The available output options are Headers Only, No History, or Full History. The default setting is Headers Only.

4.   Click out of the field or press **ENTER** to save the change.

MGCP GATEWAY PORT

Applies to MGCP gateway trunks only. The MGCP Gateway Port option defines the port number the system uses to access the MGCP gateway.

To change the port number:

1.   Select System – Devices and Feature Codes – Trunks – <***MGCP Gateway and Phone trunk***>.

2.   In the **Value** column, type or select the new port number. The valid port range is 1024–65535; the default is 2427.

3.   Click out of the field or press **ENTER** to save the change.

COMMUNICATION TIMEOUT

Applies to MGCP gateway trunks only. The Communication Timeout option specifies the number of seconds that the gateway waits for an acknowledgement from an associated phone before it considers the phone offline.

To change the Communication Timeout option:

1.   Select System – Devices and Feature Codes – Trunks – <***MGCP Gateway and Phone trunk***>.

2.   In the **Value** column, select the new value from the list. The valid range is 3–300; the default value is 15.

3.   Click out of the field or press **ENTER** to save the change.

MANUFACTURER

Applies to MGCP gateway trunks only. The Manufacturer option identifies the brand of the MGCP gateway that is connected to the system. The values are None and Audiocodes MP10X, where "X" is a number.

To change the manufacturer/model:

1. Select System – Devices and Feature Codes – Trunks – <***MGCP Gateway and Phone trunk***>.

2. In the **Value** column, select either None or Audiocodes MP10X.

3. Click out of the field or press **ENTER** to save the change.

GATEWAY NAME AND PHONE NAME

Applies to MGCP gateways and phones only. The gateway name and phone name are the names that were assigned to the gateway and associated phone when the circuits were programmed. If desired, you can change the names here.

To change a name:

1. Select System – Devices and Feature Codes – Trunks – <***trunk number***>.

2. In the **Value** column, click the current value. The Edit Gateway (or Phone) Name dialog box appears. Enter a new name, up to 18 characters, that identifies the gateway or phone on the network.

3. Click **OK** to save the change.

ASSOCIATED GATEWAY

Applies to MGCP phones only. The Associated Gateway option displays the MGCP gateway to which the MGCP phone is connected. If a gateway was not assigned when the circuit was programmed, this defaults to the first gateway.

To change the associated gateway:

1. Select System – Devices and Feature Codes – Trunks – <***trunk number***>.

2. Right-click the current extension, and then select **Change Associated Gateway**.

3. Select **MGCP Gateway and Phone**, and then click **Next**.

4. Select the desired gateway, and then click **Finish**. If you know the trunk extension for the associated gateway, you can click the current value, and then enter the extension.

5. Click out of the field or press **ENTER** to save the change.

TRUNK-RELATED INFORMATION

This section describes the following trunk-related information:

- **Call Routing Tables** below

- Loop Loss Measurement Test on **page 261**

- Music-On-Hold Profiles on **page 264**

- Loop Start AC Impedance on **page 267**

- ISDN PRI Two B-Channel Transfer on **page 268**

For information about the SIP Gateways field, see **page 230**.

CALL ROUTING TABLES

This section contains the following information:

- **Feature Description** below
- Programming Call Routing Tablesthe figure below on **page 254**
- Programming Call Routing Patterns on **page 255**

FEATURE DESCRIPTION

The system uses call routing tables (CRTs) to analyze the digits received from the following:

- Direct Inward Dialing (DID) [DDI]: Allows system users to call an system extension without dialing directly into the telephone system.

- Automatic Number Identification (ANI): Transmits the billing number, rather than the telephone number, of the calling party.

- Dialed Number Identification Service (DNIS): Provides the "800" or "900" number that callers dial to reach system.

- Caller ID [CLID]: Transmits callers' numbers to system equipment during the ringing signal, or when the call is being set up but before the call is answered.

A trunk group can be programmed to ring in to a call routing table to use the information received from DID DNIS, ANI, and Caller ID. This added information allows the phone user to receive information about the callers—such as location, name, or which advertisement they saw—on the phone display.

The system uses these digits to determine which route to use for the call, as determined by the call routing table. The caller information is passed on to the system users' display along with other call routing table information about the callers (such as location, name, or advertisement information).

Call routing tables include the following:

- **Direct Inward Dialing (DID) [DDI]**: Sends a string of digits to indicate the desired destination and includes E&M trunks. Each DID digit string can be programmed to ring in to a different destination and display a different name.

- **Automatic Number Identification (ANI)**: Transmits the billing number, rather than the telephone number, of the calling party.

- **Dialed Number Identification Service (DNIS)**: Provides the "800" or "900" number that callers dial to reach system.

- **Caller ID [CLID]**: Transmits callers' numbers to system equipment during the ringing signal, or when the call is being set up but before the call is answered.

- **ANI/Caller ID**: Identifies the source of the call. Each incoming call can be identified and sent to selected destinations. For example, calls from specific regions of the country can be sent to the appropriate sales representative or calls from specific telephone numbers can be sent to selected individuals.

- **DNIS**: Identifies the number that was dialed to reach your location. Using this number, the call routing table can send the call to the desired destination. For example, responses to advertisements using one 800 number can be sent to one hunt group and calls from other 800 numbers can be sent to other hunt groups. Or, they can all be sent to the same destination with different identifying names for the phone displays (such as Magazine Ad, TV Ad, and so forth).

A call routing table consists of fields for dialing patterns strings, ring-in types, ring-in destinations, and names. The fields are used for identifying and analyzing the source of the call and directing it to the proper destination.

- **Call routing key**: Determines which information the call routing table will use to direct the call: Trunk Number or Outside Party Number. If you select Trunk Number, the system will look for the dialed number provided by DID or DNIS. If you choose **Outside Party Number**, the system checks for the ANI or Caller ID information that identifies the source of the call.

- **Pattern strings**: Determines call routing based on a comparison of the pattern strings in the table against the digits received from the trunk interface for the incoming call. If the digits match one of the pattern strings in the table, the call is routed according to the ring-in type and destination fields associated with that pattern. If a match is not found, the call is sent to the primary attendant. However, there are two wildcard patterns that allow any number (+) or an empty pattern to match (E).

- **Ring-in type and destination**: Identifies the call routing destination and ring-in type. The call can ring in to a single extension number (phone, hunt group, Voice Processing application, individual trunk, trunk group, modem, or ARS), an extension list, another call routing table, a destination based on collected digits, or as a DISA call (which allows the caller to select a destination).

> **NOTE** When programming a call routing table that rings in to another call routing table, make sure the two call routing tables do not send calls to each other, creating a "loop."

- **Name**: Determines how the call will be identified at display phones. It can contain up to 12 characters. The name can be a word or number that identifies the call source. If the name is left blank, the system will use the priority list shown on **page 174** to determine what will be displayed on the phone.

The system can have up to 15 call routing tables. There can be a total of 900 patterns in all of the call routing tables combined.

Each individual trunk, not the trunk group, is programmed to collect digits using Caller ID, DID, ANI, or DNIS. If the trunk is programmed to collect digits, but the trunk group does not use call routing tables, the system routes the call based on the trunk group programming and ignores the collected digits. However, the collected digits will appear in SMDR and can be used by the Desktop Interface.

If the trunk is not set up to collect digits and the trunk group uses call routing tables, the system uses the call routing destination for "no-digit" calls.

CALL ROUTING ON A NETWORK

Call routing will operate as follows on a Mitel system network:

- You cannot route a call from one Call Routing Table to a Call Routing Table on another node. When a Call Routing Table sends a call to another node, it cannot be routed to any additional nodes.

- Trunks that ring in to call routing tables can only ring in to tables that are located on their local node. They cannot ring in to a call routing table on another node.

- The network attempts to optimize routes so that calls do not route through a node more than one time (looping). However, the network does not have the ability to dynamically reroute. That is, if a node trunk group that was previously unavailable becomes available after the network has already chosen another route, the network does not reroute using that node trunk group.

- The network will try to optimize routes when calls have been transferred or reverse transferred across several nodes, which may form loops.

Several sample call routing tables are shown on the following pages. These lists are abbreviated to show how the tables work. Actual tables would contain more than the sample entries.

DID Call Routing Table Example

The following is an example of a DID routing. (An actual DID table would have more than 4 entries; this is an abbreviated list.) Each DID number is identified by the 3 digits that are sent by the central office (000-003). The base number of the DID numbers is 9619. Therefore, the names of each of the patterns combine the base number and the additional 3 digits to form the complete number for display (961-9000 to 961-9003).

PATTERN	RING-IN TYPE	NAME	RING-IN DESTINATION
000	Single	961-9000	Extension Number 1001
001	DISA	961-9001	Not Applicable
002	Extension List	961-9002	Extension List P004
003	Single	961-9003	Hunt Group Number 2550

EXAMPLE #1

INCOMING CALL: Caller dials 961-9000 → CO → DID TRUNKS: Digits 000 sent to system → SYSTEM CALL ROUTING TABLE: 000 = Ext. 1001 → DISPLAY: 961-9000 RINGING IN — Ext. 1001
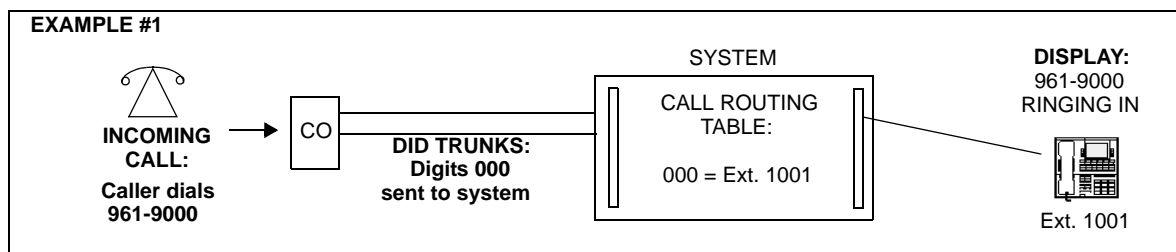
ANI Call Routing Table Example

The following is an example of an ANI table for various area codes. If the incoming ANI information begins with one of the indicated area codes, it will be routed to the designated destination and the display will show the state where the call was placed.

PATTERN	RING-IN TYPE	NAME	RING-IN DESTINATION
602+	Single	Arizona	Extension Number 2001
505+	DISA	New Mexico	Not Applicable
801+	Extension List	Utah	Extension List P005
904+	Single	Florida	Hunt Group Number 2560

EXAMPLE #2

Caller dials 602-961-9000 → CO → 602+ = ARIZONA TO SYSTEM: 602-961-9000 → SYSTEM CALL ROUTING TABLE: → DISPLAY: ARIZONA RINGING IN — Ext. 2001
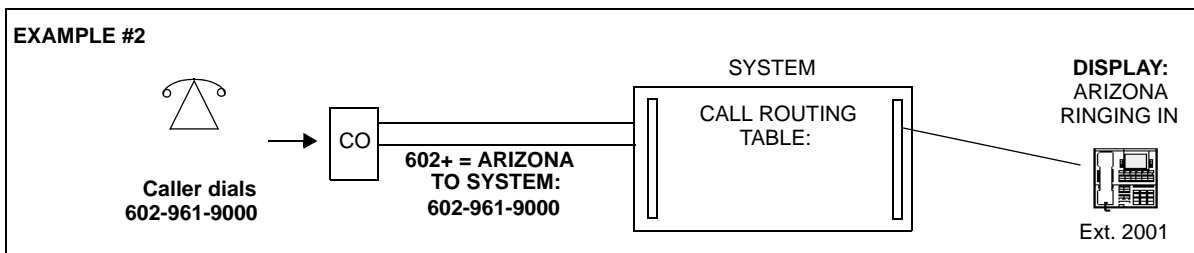
Caller ID Call Routing Table Example

The following is example table is set up for Caller ID. It is like the ANI table except that it uses office codes instead of just area codes, and it includes 1 entry for a complete telephone number. However, if the phone caller information flags are enabled, the Caller ID information received from the CO overrides the call routing table's name on the phone display. See the priority list shown on .

PATTERN	RING-IN TYPE	NAME	RING-IN DESTINATION
602961+	Single	Chandler	Hunt Group Number 2570
602893+	DISA	Phoenix	Not Applicable
602255+	Extension List	Phoenix	Extension List P006
6022549400	Single	Phoenix Branch	Extension Number 2030

EXAMPLE #3

INCOMING CALL: Call from 602-961-9000 → CO → CALLER ID DIGITS SENT TO SYSTEM: 602-961-9000 → SYSTEM CALL ROUTING TABLE: → DISPLAY: CHANDLER RINGING IN — HUNT GROUP Ext. 2570
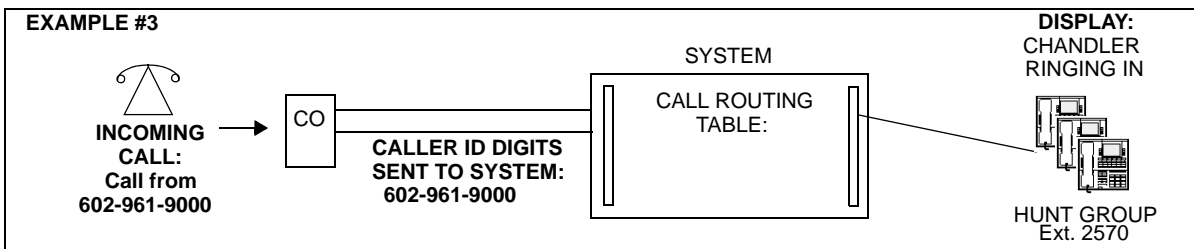
DNIS Call Routing Table Example

This is an example of a DNIS application pattern. If the DNIS digits match 8006695858, the call came in on the Technical Support 800 number and is sent to the Technical Help hunt group. The display will show that the call is from "Help Line."

PATTERN	RING-IN TYPE	NAME	RING-IN DESTINATION
8006695858	Single	Help Line	Hunt Group Number 2001
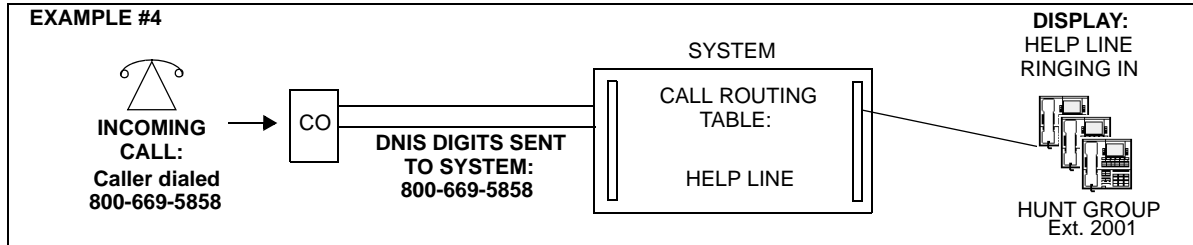
Ring-in Destination Example

This table has a ring-in destination that sends calls to another call routing table. It also uses two-stage caller identification (*ANI*DNIS*). In this example, a caller from area code 520 who dials the number (602) 961-9000 would be routed to the AZ Hunt Group, as shown in the following table.

PATTERN	RING-IN TYPE	NAME	RING-IN DESTINATION
520+	Call Routing Table	Table 2	Call Routing Table 02

If the caller in the 520 area code dialed any other number in area code 602, the call would be routed to extension 1002, as shown in the following table.

PATTERN	RING-IN TYPE	NAME	RING-IN DESTINATION
6029619000	Single	AZ Hunt Group	Hunt Group Number 2580
602+	Single	Arizona	Extension Number 1002
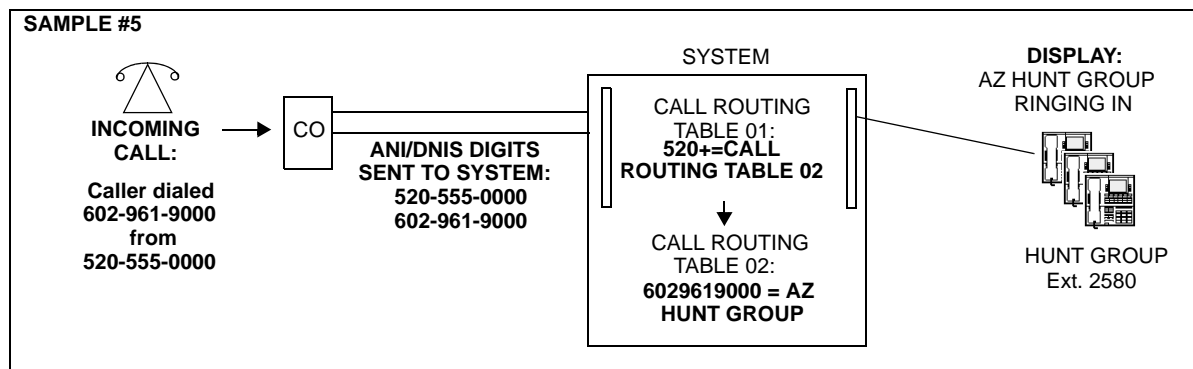
Public Network Call Routing Example

For call routing to the public network, this table is programmed to send callers who dial a special number (940-1431) to an outside line.

PATTERN	RING-IN TYPE	NAME	RING-IN DESTINATION
6029401431	Single	CR To Public	Trunk Group 92001
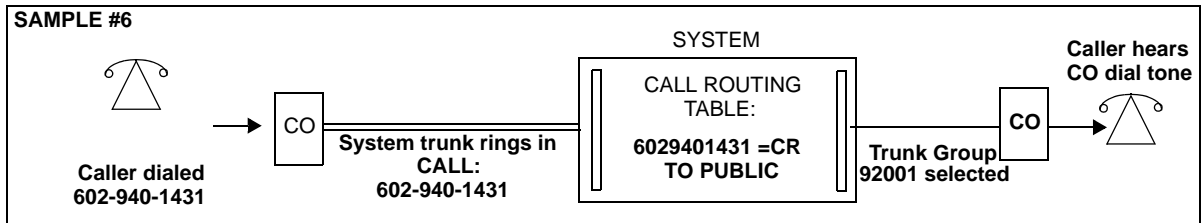
Call Routing to Another System Network Example

For call routing to another node in the system network, this table is programmed to send callers who call Node 1 using 602-961-9000 to a hunt group on Node 2. It also identifies the call as being from the "Sales Line."

NODE 1/TABLE 01

PATTERN	RING-IN TYPE	NAME	RING-IN DESTINATION
6029619000	Single	NODE 2	Node 02

NODE 2/TABLE 01

PATTERN	RING-IN TYPE	NAME	RING-IN DESTINATION
6029619000	Single	SALES LINE	Hunt Group Number 2501
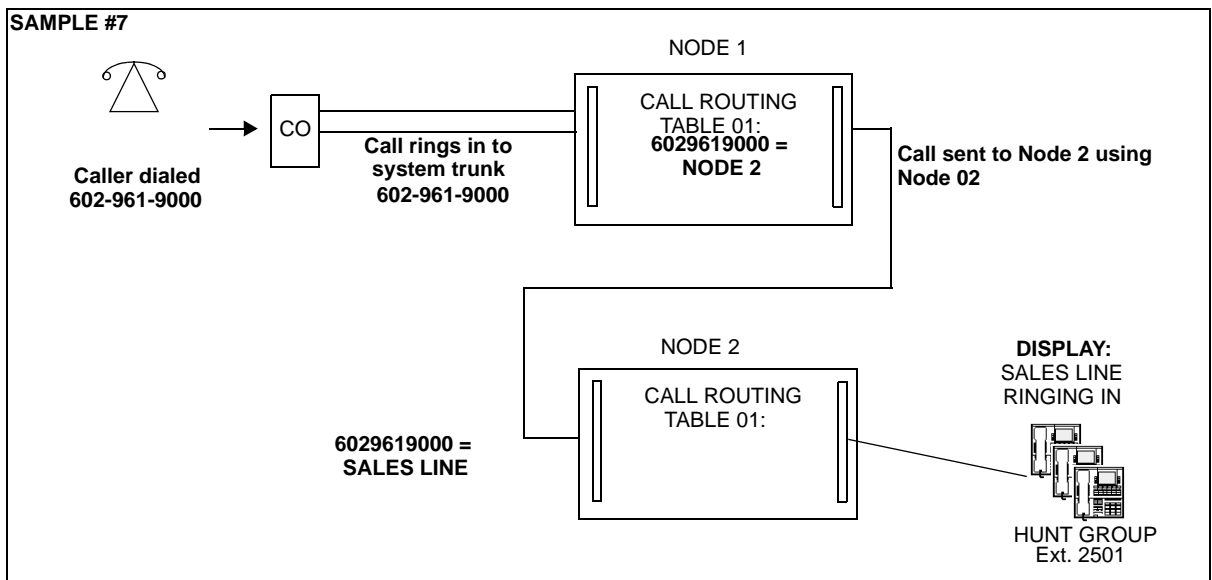
PROGRAMMING CALL ROUTING TABLES the figure below

The figure below shows the DB Programming Call Routing location.
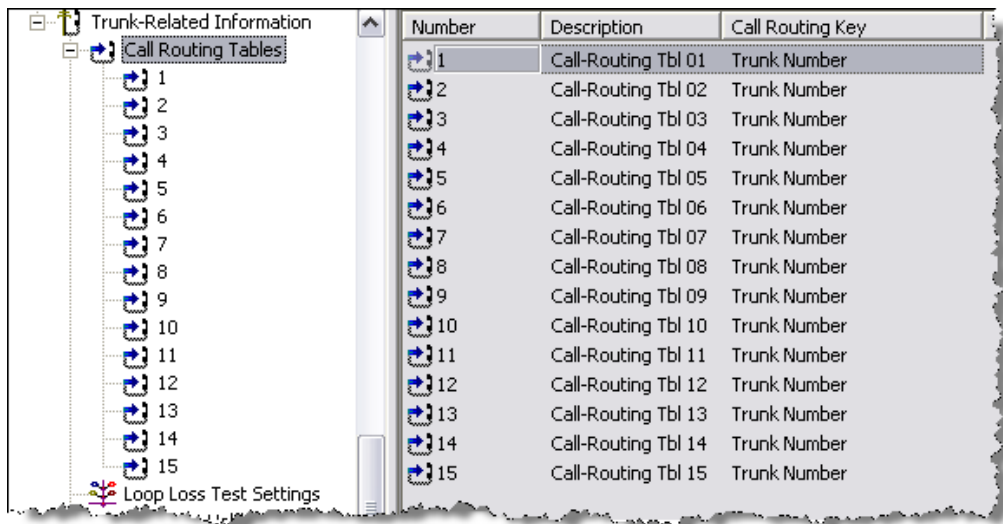
A call routing table consists of fields for dialing patterns, ring-in types/destinations, and names. The fields are used for analyzing the source or destination of the call and directing it to the proper destination.

Each individual trunk, not the trunk group, is programmed to collect digits using Caller ID, DID, ANI, or DNIS (CLID or DDI in Europe). If the trunk is programmed to collect digits, but the trunk group does not use call routing tables, the system routes the call based on the trunk group programming and ignores the collected digits. If the trunk is not set up to collect digits and the trunk group uses call routing tables, the system uses the call routing destination for "no-digit" calls. (See **page 160** for trunk programming information.)

Call Routing Tables allow patterns that route calls to destinations of individual trunks, trunk groups, and ARS anywhere single ring-in destinations are programmed.

If the caller is routed to a trunk or trunk group that is busy, the system will camp on to the trunk. The caller will hear busy tones, followed by music. When a trunk becomes available, the caller hears dial tone and can complete the call.

To provide security on outgoing calls made through direct trunk-to-trunk interfaces, the toll restriction of the selected *outgoing* trunk will be checked. However, if ARS is used, toll restriction is not checked. The toll restriction of the *incoming* trunk group and that trunk "Subject To Toll Restriction" flag are not checked. If a caller dials a number that is not allowed through toll restriction, the call will be routed to the primary attendant.

IMPORTANT	Even though Primary Rate trunks appear in the selection lists, they can only be used by selecting ARS; individual B-channel trunks or trunk groups containing B-channel trunks do not function properly with this feature.

CHANGING CALL ROUTING TABLE DESCRIPTIONS

You can change CRT descriptions.

To program Call Routing Descriptions:

1. Select System – Trunk-Related Information – **Call Routing Tables**. The Call Routing Table list (1–15) appears.

2. In the **Description** column, select the current value, and then type the new description in the box.

3. Click out of the field or press **ENTER** to save the change.

PROGRAMMING CALL ROUTING KEYS

Call Routing Keys determine which information the CRT uses to direct the call.

To program Call Routing Keys:

1. Select System – Trunk-Related Information – **Call Routing Tables**. The Call Routing Table list (1–15) appears.

2. Select the current value, and then select one of the following options:

    •   **Trunk number**: The system looks for the dialed number provided by DID [DID] or DNIS.

    •   **Outside Party Number**: The system checks for the ANI or Caller ID [CLID] information that identifies the source of the call.

3. Click out of the field or press **ENTER** to save the change.

PROGRAMMING CALL ROUTING PATTERNS

CRT patterns are listed in the CRT in the order they are checked against the incoming digits received from the trunk interface for the incoming call. When the system finds a match with a pattern, it stops searching through the list and routes the call according to the programming for that pattern. If the call reaches the end of the list without matching any patterns, the call is automatically sent to the primary attendant. Pattern entries can include up to 32 digits. **Table 44** shows valid characters for CRT entries.

NOTE	To ensure all patterns are included in the CRT, make sure that the last two patterns always include + followed by E. The + pattern allows the system to detect any digits that did not match the other patterns in the list. The E pattern allows the system to detect when no digits are received. This ensures that the call is labeled as either ANY or EMPTY when the call is sent to the primary attendant. These labels notify the attendant that the Telco did not properly route the DNIS digits.

Table 44 shows Call Routing Table Pattern characters.

Table 44: Call Routing Table Pattern Characters

CHARACTER	MATCHES
0–9,*, #	Specific digits
X	Any digit 0–9
A	Any digit 0–9, #, or *
N	Any digit 2–9
Z	Digits 0 or 1
[x]	Optional digits; for example, 2[13]2 matches 22 or 2132.
(x-x)	Range of digits; for example, with a U.S. system, (320–360)XXXX matches any 7-digit number with an office code between 320 and 360, inclusive; with a European system, (200–209)XXXX would catch all 7-digit DDI numbers in the range 200–209.
+	Matches 0 or more digits of any value. A "+" by itself will match any incoming information, including anonymous calls that send no digits.
E	End of dialing; the pattern will not match if any digits are dialed beyond this point. An "E" by itself in a pattern will match when no digits are received, such as anonymous Caller ID [CLID] calls.

EDITING CALL ROUTING TABLE PATTERNS

You can edit CRT patterns.

To edit call routing patterns:

1. Select System – Trunk-Related Information – **Call Routing Tables**. The Call Routing Table list (1–15) appears.

2. Double-click the CRT. A list of existing patterns, if any, appears in the right pane.

3. Program the following options as necessary.

    • **Description**: Determines how the call is identified at display phones when received. The name can be any word or number (up to 12 characters) that identifies the call source. You can use any combination of letters and numbers.

      To change the description of the pattern:

      1) Select the current value, then enter the new description in the text box.
    1. Press **ENTER** or select another field to save the change.

    • **Ring-In Type**: The following ring-in types are available:

      • **Single**: The destination can be a phone, hunt group, application, trunk group, individual trunk, or ARS. These can be local or off-node devices.

      • **Extension List**: You can select from any extension list in the local node.

      • **Call Routing Table**: When programming a CRT that rings in to another table, make sure the two tables do not send calls to each other, creating a "loop." You can only select call routing tables located on the local node.

      • **DISA**: This allows the caller to select a destination.

- **Collected Digits**: This ring-in type indicates that the collected DID [DDI] or DNIS digits (plus the base digits) should be used as the destination extension. This variable helps to keep the number of call routing table entries to a minimum.

  Both trunk groups and call routing table entries can use this ring-in type. When Ring-In Type of Collected Digits is selected, the Ring-In Destination field is empty. If the collected digits plus the base digits do not make up a valid ring-in destination, the call is routed to the primary attendant. Valid ring-in destinations include on- or off-node phones, on- or off-node hunt groups, trunk groups, individual trunks, voice mail applications, automated attendants, and ARS.

> **NOTE** If Single, Extension List, or Call Routing Table is selected, you must also program the destination as described below. DISA ring-in requires no additional programming. Any DISA security codes programmed for the trunk group (see **page 521**) will apply to callers.

To change the Ring-In Type:

1) Select the current Value, then scroll to the desired setting.

2. Press **ENTER** or select another field to save the change.

- **Ring-In Destination**: To select the specific destination for the pattern, use one of the following methods:

Method A

1) Select the current value, and then type the new value in the text box.

3. Press **ENTER**. A screen appears displaying what is associated with the number entered.

4. Click **OK**. The new number appears in the field.

Method B

1) Right-click the existing Ring In Destination. An option box appears.

5. Select the Change Ring-In Destination option. A window appears prompting for the device type to include.

6. Select the appropriate destination type, and then click **Next**. The list of devices appears. You can view them in a list by selecting the List button or view details by selecting the Details button. If the Ring-In Destination is set to "None" the call will be sent to the primary attendant. Do not set this to None if there is not a primary attendant, because the call will not ring at any phone. Even though Primary Rate trunks appear in the selection list, they can only be used by selecting ARS; individual B-channel trunks or trunk groups containing B-channel trunks will not function properly with this feature.

7. Select the desired device, then click **Finish**. The selection appears in the Ring-In Destination field.

VIEWING CALL ROUTING PATTERNS

To program the patterns for the Call Routing Table:

1. Select System – Trunk-Related Information – **Call Routing Tables**. The Call Routing Table list (1–15) appears.

2. Double-click the Call Routing Table. A list of existing patterns, if any, appears in the right pane.

DELETING A PATTERN

To delete a pattern:

1. Select System – Trunk-Related Information – **Call Routing Tables**. The Call Routing Table list (1–15) appears.

2. Double-click the Call Routing Table. A list of existing patterns, if any, appears in the right pane.

3. Select the pattern you want to delete, right-click, and then select **Remove Selected Items**.

ADDING A SINGLE PATTERN

To add a single pattern:

1. Select System – Trunk-Related Information – **Call Routing Tables**. The Call Routing Table list (1–15) appears.

2. Double-click the Call Routing Table.

3. Right-click anywhere in the right pane, and then select **Add To List**. The new pattern appears in the list.

COPYING AND PASTING PATTERNS

To copy and paste one or more patterns:

1. Select System – Trunk-Related Information – **Call Routing Tables**. The Call Routing Table list (1–15) appears.

2. Double-click the Call Routing Table.

3. Select the pattern that you want to copy.

4. Right-click, and then select **Copy Selected Pattern to**. The Copy Pattern(s) dialog box appears displaying the Current CRT.

5. Select the Destination CRT using the drop-down list box.

6. Click **Copy** to copy the pattern(s) from the Current CRT to the Destination CRT.

SELECTING BATCH CREATE PATTERNS

You can use the Batch Create Patterns feature for either U.S. or European installations.

To Batch Create Patterns:

1. Select System – Trunk-Related Information – **Call Routing Tables**. The Call Routing Table list (1–15) appears.

2. Double-click the Call Routing Table.

3. Right-click anywhere in the right pane, and then select **Batch Create Patterns**.

4. Select **Area Code** or **DID [DDI]**, and then click **Next**. A dialog box appears to program the pattern options.

5. To program an area code, follow the steps below. To program a DID number, go to **step 6**.

    a. In the **Pattern Name** box, type the name. The name is used for all area code patterns created using this batch command and determines how calls are identified at display phones. The name can be a word or number that identifies the call source. You can use any combination of letters and numbers.

    b. In the **Ring-in Type** area, select the Ring-in Type option for the Call Routing Pattern. The ring-in type is used for all area code patterns created using this batch command. Ring-in destination types are as follows (see **page 256** for descriptions):

       • Single extension number (off-node device, phone, hunt group, application, trunk group, individual trunk, or ARS). Also requires the Ring-in Destination (see **step c**).

       • Extension list. Also requires the Ring-in Destination (see **step c**).

       • DISA (allows the caller to select a destination)

       • Collected digits

       • Call routing table. Also requires the Ring-in Destination (see **step c**).

    c. *For Single, Extension List, and Call Routing Table Ring-in types only (see* **step b***).* Click **Ring-In Destination**, and then select the destination type. After you return to the Call Routing Name and Ring-In Information screen, click **Next** to continue.

> **NOTE** Even though PRI trunks appear in the selection lists, they can only be used by selecting ARS; individual B-channel trunks or trunk groups containing B-channel trunks will not function properly with this feature.

    d. Select the area codes that will be included in the batch of patterns. (You can use the SHIFT or CTRL key to select more than one area code.)

    e. Click **Finish**.

6. Program the following DID [DDI] options:

U.S. DID installations only.

    a. In the **Enter the Description** box, type the base number of the DID number block in the box. Each pattern is identified in the list by the digits that are sent after the hand-shake. The base number of the DID numbers is the name prefix. The prefix and the collected digits are combined to form a complete number for display (for example, a base number 9619 would combine with digits 000 to display 9619000).

    b. In the **Enter the Initial DID** box, type the first number in the series of DID numbers being programmed. For example, if you were programming for DID numbers 9619000 through 9619019 (9619 is the base number), you would type 000.

    c. In the **Enter the Number of Patterns** list, select the number of DID numbers (up to 999) to include in the pattern list. In the example above (9619000 through 9619019) you would enter 20 because the range includes 20 DID numbers. Scroll to or enter the appropriate number (up to 999) for your DID list in the box.

    d. Click **Finish**.

European DDI installations only.

a. In the **Enter the Description** box, type the base number of the DDI number block in the box. Each pattern is identified in the list by the digits that are sent after the handshake. The base number of the DDI numbers is the name prefix. The prefix and the collected digits are combined to form a complete number for display (for example, a base number 01162903 would combine with digits 000 to display 01162903000).

b. In the **Enter the Initial DDI** box, type the first number in the series of DDI numbers being programmed (without the base number). For example, if you were programming for DDI numbers 01162903000 to 01162903019 (01162903 is the base number), you would enter 000.

c. In the **Enter the Number of Patterns** list, select the number of DDI numbers (up to 999) to include in the pattern list. In the previous example (01162903000 to 01162903019), you would enter 20 because the range includes 20 DDI numbers.

d. Click **Finish**.

PERSISTENT MUSIC-ON-HOLD SELECTION

You can route CO trunk calls through CRTs and use the matching CRT entry's "Music-On-Hold Profile" to overwrite the settings at the CO trunk group. The Music-On-Hold profile provides the same Music-On-Hold fields that currently exist in a CO trunk group. See **page 264** for details.

LOOP LOSS MEASUREMENT TEST

This section contains the following information:

- **Feature Description** below

FEATURE DESCRIPTION

One of the utilities that now resides within the MiVoice Office 250 System Administration and Diagnostics application. The test results appear in Message Print, as shown below. For more information, refer to the *MiVoice Office 250 System Administration and Diagnostics Help*.

CONFIGURING LOOP LOSS MEASUREMENT TEST FIELDS

To configure global settings for the loop loss measurement test:

1. Select System – Trunk-Related Information – **Loop Loss Test Settings**.

2. Program the following fields:

   - **Test Number**: Type the phone number to call that plays the test tone. Do not prepend an outgoing digit. The range is 15 numeric characters.

- **Test Transmit Level in dBm**: The Test Transmit Level is the volume of the tone that the test number sends when you call it. This tells the system what the original tone was, and it compares the tone that it measures against that to calculate the loop loss. The value needs to come from whoever you get the test number itself from (that is, whoever provided your analog trunks). Usually this field is either 0 or -10 dBm (most likely 0 dBm). The range is 1–120 dBm; the default is 0 dBm.

- **Warning Threshold for Loss in dB**: Specifies how much loss must be present for Call Processing to issue a Message Print indicating that there has been too much loss on the line. The range is 0–75 dB; the default is 8 dB.

- **Number of Test Passes**: The length of time after you dial the number before the DSP Resource starts measuring the signal. You must time how long it takes the tone to start playing from the time you dial the last digit of the test number. The range is 0–255 ms; the default is 10 ms.

- **Duration of Measurement in milliseconds**: Specifies how long the DSP Resource spends measuring the signal. The range is 1-65535 ms; the default is 500 ms.

STARTING A LOOP LOSS MEASUREMENT TEST

To start a Loop Loss Measurement Test on a single trunk:

1. Select System – Devices and Feature Codes – **Trunks**.

2. In the **Extension** column, right-click on the trunk, and then select **Run Loop Loss Test**, as shown below. The test results appear in Message Print.
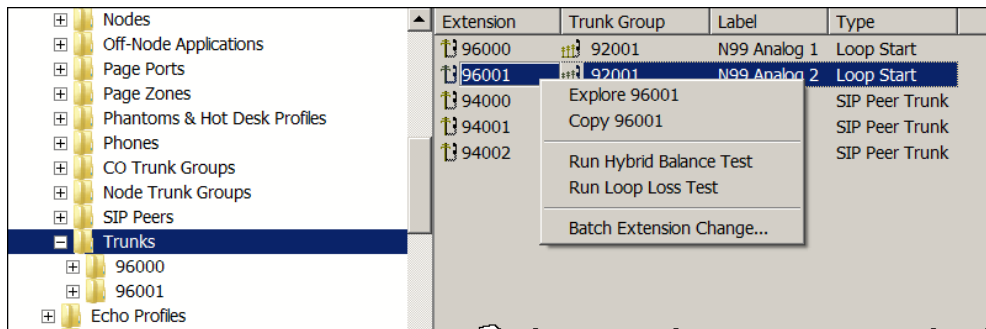
TROUBLESHOOTING THE LOOP LOSS MEASUREMENT TEST

Table 45 lists troubleshooting information for the Loop Loss Measurement Test feature.

Table 45:   Loop Loss Measurement Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Results are not consistent with the test equipment.	The fields are not programmed correctly in DB Programming.	Add more test passes to adjust the timing or change the delay in DB Programming.
	The local test number and transmit level of the CO is not the same as what is configured in DB Programming.	Confirm the local test number and transmit level of the CO. Make sure DB Programming matches the CO.

MUSIC-ON-HOLD PROFILES

This section contains the following information:

- **Feature Description** below

- Programming Music-On-Hold Profiles for CRTs on **page 265**

- Troubleshooting Music-On-Hold for CRTs on **page 266**

FEATURE DESCRIPTION

MiVoice Office 250 Call Processing manages the audio connections for incoming Public Switched Telephone Network (PSTN) calls. When a device places a call on hold, CO trunk group settings determine the hold audio (see "Programming Trunk Group Options" on **page 506**.)

This determination is based on the settings at the Central Office (CO) trunk group to which the trunk resides. The settings include "Music-On-Hold," "Audio on Transfer to Ring," "Audio on Transfer to Hold," and "Audio on Hold for Transfer Announcement." Each of these fields allows for various options, including Silence, Tick Tone, Ringback, Board, Music-On-Hold, Use Next Device's Audio Source, and File-Based MOH.

You can route CO trunk calls through CRTs and use the matching CRT entry's "Music-On-Hold Profile" to overwrite the settings at the CO trunk group. The Music-On-Hold profile provides the same Music-On-Hold fields that currently exist in a CO trunk group.

Table 46 shows an example of how the Music-On-Hold Profiles are handled when three CRT entries are chained.

Table 46: An Example of the Music-On-Hold (MoH) Profiles for Chained CRTs

	CRT #1	CRT #2	CRT #3	RESULT
MoH Profile IDs	None	None	None	Use the CO Trunk Group's MoH settings
	1	None	None	Use MoH Profile ID 1
	1	2	3	Use MoH Profile ID 3

If the matching CRT entry does not have a "Music-On-Hold Profile" set or the call is not routed through a CRT, the trunk determines the audio setting based on the CO trunk group settings. Therefore, the "Music-On-Hold Profile" in the matching CRT entry has priority over the audio settings in the CO trunk group.

If CRT entries are "chained" (a CRT entry may point to another CRT), the last matching CRT entry in the chain with a Music-On-Hold Profile set to anything other than "None" takes precedence. If the last matching CRT entry has a Music-On-Hold Profile set to "None," the associated CO trunk group settings apply.

When the Music-On-Hold selection is set for a specific call, the Music-On-Hold setting persists through the duration of the call even if that call is forwarded, moved, or transferred.

The following are usage examples:

- Several businesses share a single phone system. Due to the layout and size of the companies involved, each business does not require its own CO trunk group. However, each business in the group owns a specific subset of the DID [DDI] numbers available. These businesses require their own Music-On-Hold played for callers. Using Persistent Music-On-Hold, each business would have a Music-On-Hold source assigned and would specify that source within the CRT entry for the matching called number.

- A business sells several different products. Customers call different numbers based on the product of interest. Management would like customers to hear Music-On-Hold based on the product in question. This feature would allow the specific CRT entry to dictate which source provides the Music-On-Hold for the incoming calls.

PROGRAMMING MUSIC-ON-HOLD PROFILES FOR CRTS

You must program Music-On-Hold Profiles before you can assign them to CRTs (see the following section).

CREATING MUSIC-ON-HOLD PROFILES

To create Music-On-Hold Profiles:

1. Select System – Trunk-Related Information – **Music-On-Hold Profiles**.

2. Right-click **Add to Music-On-Hold Profiles List** in the right pane. The Get ID dialog box appears.

3. Select the starting ID and the number of IDs (up to 25) to create. For example, to create IDs 1–9, select **1** as the "Starting ID" and **9** as the "Number of IDs."

4. Click **OK**. The items are added to the list with default values.

5. In the **Description** column, type the profile description.

6. Double-click the Music-On-Hold Profile that you want to program.

7. For each profile, select one of the following options from the list:
   - Music-On-Hold. For a feature description, see "Music-On-Hold" on **page 514**.
   - Audio on Transfer to Ring. For a feature description, see "Audio on Transfer To Ring" on **page 515**.
   - Audio on Hold for Transfer Announcement. For a feature description, see "Audio On Hold For Transfer Announcement" on **page 516**.
   - Audio on Transfer to Hold. For a feature description, see "Audio On Transfer To Hold" on **page 515**.

8. Click out of the field or press **ENTER** to save your change.

ASSIGNING MUSIC-ON-HOLD PROFILES TO CRTS

To assign Music-On-Hold Profiles to CRTs:

1. Select System – Trunk-Related Information – **Call Routing Tables**. A list of CRTs appears.

2. Click the desired CRT.

3. Right-click anywhere in the right pane, and then click **Add To List**. The Patterns list appears.

4. Do one of the following:

    • Type the desired profile ID in the **Music-On-Hold Profile** field.

    • Select the profile ID from the Change Music-On-Hold Profile dialog box:

        a. Right-click the existing Music-On-Hold Profile, and then select **Change Music-On-Hold Profile**. The Change Music-On-Hold Profile dialog box appears.

        b. Select **Music-On-Hold Profiles**, and then click **Next**.

        c. Select the desired Music-On-Hold profile, and then click **Finish**.

TROUBLESHOOTING MUSIC-ON-HOLD FOR CRTS

Table 47 contains troubleshooting information for Music-On-Hold for Call Routing Tables.

Table 47:   Persistent Music-On-Hold Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
No Music-On-Hold (external music source connected).	The external music source is turned off or inoperative.	Check the external music source for proper operation.
	There is a programming error.	Make sure the following fields are programmed correctly: • Music-On-Hold settings for CO Trunk Groups • Music-On-Hold Profiles for CRTs
	There is a defective cable between the music source and the Music-On-Hold port on the back of the chassis.	Repair or replace the cable. Check to see that a 1/8-inch, 2-conductor (mono for CS5000, stereo or mono for HX Controller), mini-phone plug was used.
	There is a defective Music-On-Hold port.	Replace the defective chassis.

LOOP START AC IMPEDANCE

The Loop Start AC Impedance setting consists of default line types specific to each country. Each country should have a default value that corresponds to the country type of the software build.

Table 48: AC Impedance Settings by Country

AC IMPEDANCE	COUNTRY
600 Ohms	Argentina, Brazil, Chile, China (data), Columbia, Ecuador, El Salvador, Guam, Hong Kong, India, Indonesia, Japan, Jordan, Kazakhstan, Kuwait, Macao, Malaysia, Mexico, Oman, Pakistan, Peru, Philippines, Russia, Saudi Arabia, Singapore, South Korea, Taiwan, Thailand, United Arab Emerites, United States, Yemen
900 Ohms	United States alternate
TBR21 270R + (750R//150 NF)	Austria, Bahrain, Belgium, Croatia, Cyprus, Czeck Republic, Denmark, Egypt, Finland, France, Germany, Greece, Hungary, Iceland, Ireland, Israel, Italy, Latvia, Lebanon, Luxembourg, Malta, Morocco, Netherlands, Nigeria, Norway, Poland, Portugal, Romania, Slovakia, Slovenia, Spain, Sweden, Switzerland, United Kingdom
220R + (820R//120NF	Australia, Bulgaria, South Africa
370R + (620R//310NF	New Zealand
320R + (1050R// 230NF)	United Kingdom (old)

To change the Loop Start Impedance setting:

1. Select System – Trunk-Related Information – **Loop Start Impedance**.

2. In the Value column, select the option from the list.

3. Click out of the field or press **ENTER** to save the change.

ISDN PRI TWO B-CHANNEL TRANSFER

This section contains the following information:

- **Feature Description** below
- Programming TBCT on **page 271**

FEATURE DESCRIPTION

The ISDN PRI Two B-Channel Transfer (TBCT) feature is an ISDN optimization service offered by the PSTN. You must purchase support for the TBCT from the CO. The TBCT feature optimizes trunk-to-trunk calls by releasing them from the PBX and connecting them through the Central Office (CO). This optimization removes the trunk-to-trunk call legs from the system and allows the system to reuse the 2 B-channels for new calls.

This feature supports the basic TBCT functionality from the *Telcordia specification GR-2865-CORE, Generic Requirements for ISDN PRI Two B-Channel Transfer, Issue 3, March 2000*. A future release may implement other 2B-Transfer methods based on other specifications.

TBCT includes the following options:

- **Enable ISDN Two B-Channel Transfer**: When selected, the TBCT feature is initiated for the ISDN port (the default is No), see **page 271**.

- **Display "T" for Two B-Channel Transferred Calls**: When selected, SMDR displays a "T" in the output when a TBCT occurs (the default is No). See "Display "T" for Two B-Channel Transferred Calls" on **page 85**.

TBCT includes basic functionality with support for the following conditions:

- Both the PBX and PSTN must support the feature.

- TBCT requires two PRI calls, which the PSTN connects before releasing them from the PBX. From an ISDN signaling perspective, one call must be an incoming or outgoing call to the PBX in the connected state (answered) and the other call must be an outgoing call from the PBX in the alerting or connected state.

- The two PRI calls involved in the TBCT may be on a single PRI port or in separate PRI ports.

- In a multi-PRI scenario, both PRI ports need to connect to the same CO and they must be in the same trunk group at the CO. The TBCT is still attempted, but it fails if this condition is not met.

- Although the PSTN may send the values of counters (Active Transfers and Available Transfers) in the TBCT messaging, the MiVoice Office 250 ignores these counters.

- The MiVoice Office 250 automatically attempts the TBCT as long as both calls are in the correct state, the PRI port (or both ports) has its corresponding TBCT flag enabled, and all the TBCT information is available to generate the TBCT request. You do not need to use a feature code to initiate a TBCT.

- If the PSTN rejects a TBCT request, the MiVoice Office 250 does not retry the TCBT request for the same calls. The MiVoice Office 250 logs the TBCT request failure and attempts a

new TBCT request for only new calls. The calls involved in the failed TBCT attempt remain connected as if the TBCT was never requested.

• Upon completion of a TBCT, the system no longer has the ability to monitor the transferred call. Even if the PSTN notifies the system when the transferred call terminates, the system ignores such notification. Because of this implementation, third-party System OAI applications (such as MiContact Center and MiCollab Client) are not able to detect that a TBCT occurred. These applications view the TBCT event as if both calls involved in the TBCT disconnected. Similarly, call cost records (such as those recorded in SMDR) terminate once the optimization occurs.

For multiple node configurations, both nodes with the PRIs involved in a TBCT must be running MiVoice Office 250 v3.0 or later software. Intermediate nodes do not have to be running the latest software. If one of the two nodes involved in the TBCT is not running the latest software, the systems do not attempt a TBCT and the calls work normally as unsupervised CO trunk-to-trunk calls.

For sample call flows for single and multiple nodes, refer to the following examples:

• Single Node, Single PRI Span Scenario below:

```
Outside Party #1                                          MiVO 250

                Incoming call on PRI Span-1              Subscriber

Outside Party #2   Outgoing call on PRI Span-1

                Request for 2B-XFR on PRI Span-1

                Success Response for 2B-XFR on PRI Span-1
                Disconnects for two calls on PRI Span-1
```

• Single Node, Multiple PRI Span Scenario below:

```
Outside Party #1                                          MiVO 250

                Incoming call on PRI Span-1              Subscriber

Outside Party #2   Outgoing call on PRI Span-2

                Request for 2B-XFR on PRI Span-2

                Success Response for 2B-XFR on PRI Span-1
       Disconnects for call on PRI Span-1 and call on PRI Span-2
```

- Multiple Node, Multiple PRI Span Scenario below:
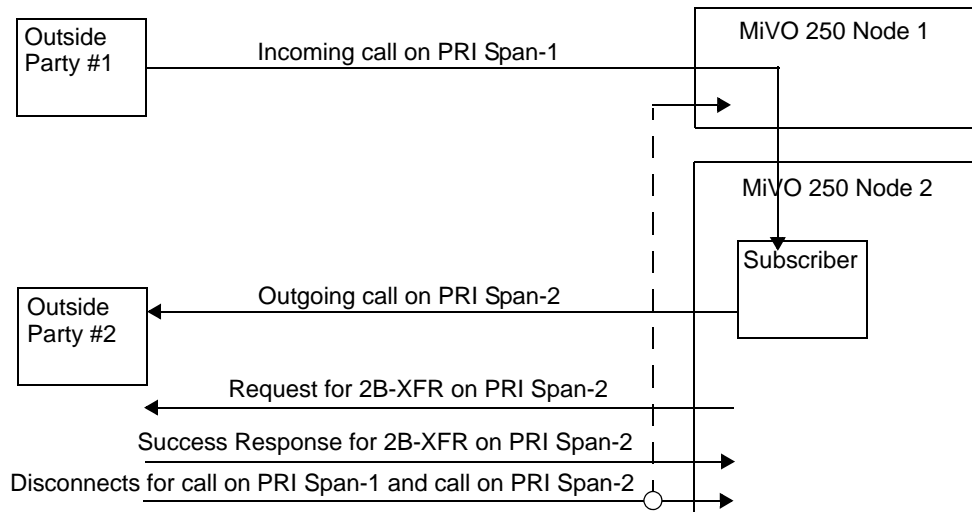
1. Outside Party #1 calls into the PBX on PRI Span-1.

2. A subscriber answers the call, and then transfers the call to Outside Party #2 using PRI Span-1.

3. A subscriber hangs up to complete the transfer. This results in an unsupervised CO trunk-to-trunk call between Outside Party #1 and Outside Party #2.

4. Because the MiVoice Office 250 knows that the trunk-to-trunk call uses the same span, the MiVoice Office 250 sends a facility request "invoke" command along with the two call reference values.

5. The PSTN acknowledges 2B-Transfer request. The PSTN disconnects both B-channel calls.

1. Outside Party #1 calls into the PBX on PRI Span-1.

2. A subscriber answers the call, and then transfers the call to Outside Party #2 using PRI Span-2.

3. A subscriber hangs up to complete the transfer. This results in an unsupervised CO trunk-to-trunk call between Outside Party #1 and Outside Party #2.

4. The MiVoice Office 250 sends a facility request "invoke" command with the D-channel IDs of PRI Span-1 along with the two call reference values.

5. The PSTN acknowledges 2B-Transfer request. The PSTN disconnects both B-channel calls.

1. Outside Party #1 calls into the PBX on PRI Span-1 on node 1.

2. A subscriber on node 2 answers the call, and then transfers the call to Outside Party #2 using PRI Span-2.

3. A subscriber hangs up to complete transfer. This results in an unsupervised CO trunk-to-trunk call between Outside Party #1 and Outside Party #2.

4. The MiVoice Office 250 sends a facility request "invoke" command with the D-channel IDs of PRI Span-1 along with the two call reference values.

5. The PSTN acknowledges 2B-Transfer request. The PSTN disconnects both B-channel calls.

PROGRAMMING TBCT

You can enable TBCT for PRI reports. To enable TBCT for Station Message Detail Recording (SMDR), see "Display "T" for Two B-Channel Transferred Calls" on .

To enable TBCT:

1. Select System – Controller – **Dual T1/E1/PRI Module** or **T1/E1/PRI Module**, and then click **T1/PRI or E1/PRI**.

2. Select **Enable ISDN Two B-Channel Transfer**.

3. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

4. Press **ENTER** or select another field to save the change.

Chapter 7

END USER FEATURES

INTRODUCTION

This chapter describes MiVoice Office 250 features that are used by or mainly affect end users. Features are listed in alphabetical order beginning on .

For voice processing features, see "Voice Processor Features and Programming" on .

For troubleshooting issues for end user features, see .

USER FEATURE INFORMATION

The following section provides information about phone user features.

ACCESS TO USER FEATURES

Each of the phone features is assigned a feature code. To select trunks, process calls, and use system features, users enter codes using the phone dialpad, feature buttons, hookflash, or Special button, as appropriate.

FEATURE CODES

Feature codes shown in this chapter are default values. You can use DB Programming to change feature codes to any 1-digit to 5-digit value.

> **NOTICE**
>
> Changing a feature code may affect the accessibility of other features.

FEATURE BUTTONS

Desktop phones provide feature buttons that allow one-button dialing of feature codes. Through DB programming and the use of keymaps, the system administrator sets up the arrangement of the feature buttons and their default values. If desired, some of the feature buttons can be designated as user-programmable buttons. For more information about keymaps, see "Keymaps" on **page 425**.

SPECIAL BUTTON

> **NOTE** Mitel IP phones use different buttons as the Special button. For more information, refer to the applicable phone user guide.

Depending on the feature and system programming, users can either enter feature codes immediately after lifting the handset or while on-hook, or they must signal the system before entering the feature code. Users signal the system by pressing the Special button. Single line DTMF phone users perform a hookflash (recall, in Europe) by quickly pressing and releasing the hookswitch. If the user does not enter a code or begin dialing before the Dial Initiation timer expires, the system sends reorder tones.

If the system-wide option "SPCL Key Required for Feature Code Entry" is enabled, users must always press the Special button before entering a feature code.

ACCOUNT CODES

Account codes are 3- to 12-digit codes that can be used in conjunction with the Station Message Detail Recording (SMDR) feature to aid record keeping. Account codes can be assigned to measure telephone use and/or to identify calls for customer billing.

See "Account Codes" on for details.

AUTOMATIC CALL ACCESS

This feature allows a user to determine the way incoming calls are answered by entering a feature code to select the type of call access.

For feature usage instructions, refer to the applicable phone user guide.

The four variations are as follows:

- **No Automatic Call Access**: The user hears intercom dial tone when the handset is lifted and must press a CALL or individual trunk button (or the ANSWER button) to answer an outside call, or press the IC button (or CALL button, if there is no **IC** button) to answer an intercom call.

- **Automatic Intercom Call Access Only**: The user can answer ringing intercom calls by simply lifting the handset, but outside calls must be answered by pressing a CALL button, an individual trunk button, the ANSWER button, or the ANSWER menu option.

- **Automatic Outside Call Access Only**: The user can answer ringing outside calls by lifting the handset, but ringing intercom calls must be answered by pressing the **IC** button (or CALL button, if there is no IC button).

- **Automatic Access For All Calls**: The user can answer any ringing call by lifting the handset. If no call is ringing, the user hears intercom dial tone when the handset is lifted. This is the way single line phones work. When the system is in the default state, all phones have this type of call access.

NOTE	In any of the preceding variations, users may still receive handsfree intercom calls (if enabled).

The selected option determines how all types of ringing intercom or outside calls (direct calls, transferred calls, recalls, and so forth) are answered. If more than one call is ringing at the phone, the first call received is the first answered.

When programmed for automatic outside call access, a phone user with allowed answer, but without ring in, for a ringing trunk must always press an individual trunk button for that trunk, or enter the Automatic Trunk Answer feature code (350) to answer the incoming call. Transferred calls and recalls can be answered by lifting the handset.

Camped-on calls cannot be answered by simply lifting the handset or pressing the Speaker button. For example, a phone is programmed to automatically answer ringing outside calls, but requires pressing the IC button to answer ringing intercom calls. If a private intercom call rings in and is immediately followed by an outside call ringing in, the display shows the intercom call message, and the outside call camps on. The intercom call also camps on when the handset is lifted. The user can then choose between the camped on calls by pressing either the IC button or the CALL or individual trunk button, ANSWER button, or ANSWER menu option.

The automatic call access options described in the following paragraphs can be programmed multiple-line phones only. Single line phones are designed to automatically answer ringing intercom and outside calls by lifting the handset and cannot be changed.

CALL LOGGING

The Call Logging feature helps you see who called when you were away and makes it easy to redial those people. Non-display and single line phones do not support this feature. Six-line display phones are recommended for field visibility and ease of use.

See "Call Logging" on **page 395** for details.

CALL SCREENING

When a call is received by an automated attendant or Call Routing Announcement application and the caller enters an extension number, the programmed Transfer Method determines how the call will be transferred. The Transfer Method flags in the database determine the methods used for transferring calls to phones with mailboxes, phones with extension IDs, and extensions without mailboxes or IDs. If allowed in mailbox programming, they can also be programmed by the mailbox user.

For feature usage instructions, refer to the applicable phone user guide.

For programming instructions, see Transfer Method on .

The available Transfer Methods include:

- **Announce Only***:* The caller is asked to state his or her name. Then the call is transferred to the associated extension number. When the phone user answers the transfer, the voice processing system plays the caller's name and completes the transfer.

- **Screened***:* The caller is asked to state his or her name. Then the call is transferred to the associated extension number. When the phone user answers the transfer, the voice processing system plays the caller's name. The phone user has the options of replaying the name, sending the call to voice mail (if the extension has a mailbox), transferring the call to another extension, accepting the call, or rejecting the call.

- **Unannounced**: The call is transferred to the associated extension number without announcing the call. This is the default method.

In a network setting, an external voice processing system can provide call screening for a destination extension on another node. However, the node where the external voice processing system is connected must have an off-node device programmed for the destination extension and access to the remote node.

Receiving an Announced or Screened Transferred Call: Depending on the Transfer Method programmed for your mailbox or extension ID, one of the following will occur when you receive a call transfer from the automated attendant or a Call Routing Announcement application:

- **"Announce Only" Calls:** When the caller enters your extension number, the system asks the caller to record his or her name. The voice processor then places a private call to your phone. When you answer, it plays *"You have a call from (caller's name)."* It then completes the transfer, and the transferred call rings at your phone. Answer as usual.

- **"Screened" Calls:** When the caller enters your extension number, the system asks the caller to record his or her name. The voice processor then places a private call to your phone. When you answer, it plays *"You have a call from (caller's name),"* and gives you the following options:

    - *To accept the call,* press **#** or the **ACCEPT** menu button. The call will ring at your phone.

    - *To replay the announcement,* press **1** or the **REPLAY** menu button.

    - *To send the call to voice mail,* press **2** or the **SEND TO V-MAIL** menu button. The caller is transferred to your mailbox and hears your personal greeting.

- • *To forward the call to another extension,* press **3** or the **FORWARD** menu button. Then enter the extension number. The call is sent to the other phone. If that phone has screened or announced transfers, the caller's name will be played again for that phone.

- • *To refuse the call,* press ***** or the **REFUSE** menu button, or simply hang up. The caller receives a recording that says you are not available and offers them the option of leaving a message.

- • **"Unannounced" Calls:** This is the default method for transferring a call. The call is transferred to your phone without any kind of announcement.

CALL WAITING

While a phone is in use, incoming intercom and outside calls camp on until the busy phone is available. The busy party hears a single camp-on tone every 15 seconds (unless the Camp On Tone timer is changed or Camp On tones are disabled at the phone).

A phone programming flag (Camp-On Indications) allows Camp On tones to be enabled or disabled for incoming intercom and outside calls. This flag is programmed on a phone-by-phone basis. See

The DID/E&M Receive Busy Instead Of Camp-On phone flag determines whether E&M and DID callers will receive busy signal or receive ringback and camp on when calling a busy phone. In the default state, busy tones are disabled, and the callers will hear ringback while camped on to the called phone.

For programming instructions, see .

For feature usage instructions, refer to the applicable phone user guide.

CONFERENCE CALLS

The following types of conference calls are available:

- **Ad Hoc Conferencing** below

- Meet-Me Conferencing on **page 283**

Refer to the *MiVoice Office 250 Installation Manual* for details about conferencing capacities.

AD HOC CONFERENCING

This section contains the following information:

- **Feature Description** below

- Programming Ad Hoc Conferencing on **page 283**

FEATURE DESCRIPTION

Phone users can establish multi-party Ad Hoc[1] conference calls without operator assistance. Prior to v5.0, the traditional Ad Hoc Conferencing feature was limited to four-party conference calls and 20 total conferencing resources, which allowed for a system maximum of five simultaneous four-party conference calls.

With v5.0 or later, Ad Hoc Conferencing can be enabled to support up to eight-party conference calls and 20 total conference resources, which allows for a system maximum of two eight-party and one four-party Ad Hoc Conference calls.

For feature usage instructions, refer to the applicable phone user guide.

During a conference, some reduction in voice volume may be noticed, depending on CO trunk quality. And, if any phone user presses a keypad button, the DTMF tones will be heard by all other parties in the conference. This allows conference callers access to DTMF-controlled devices.

An established conference can be transferred to another phone, using the call transfer feature described in "Transfer – Call Transfer" on **page 337**. While the transfer is taking place, the parties in the conference remain connected to each other and may converse. The transfer will appear at the destination phone in the same manner as any other transferred call along with a CONFERENCE TFR FROM <USERNAME> display and may be answered by the party.

> **NOTICE**
>
> While this system is designed to be reasonably secure against CO trunk misuse by outside callers, there is no implied warranty that the system is not vulnerable to unauthorized intrusions. If the telephone company CO does not provide line supervision and does not disconnect the call when one party hangs up, it is possible for a caller to remain connected to a CO trunk circuit. If this should happen and the caller begins dialing, the call could be placed through the Mitel system and would then be billed to the system's owner. The system cannot check this type of call for toll restriction and may not register the call in SMDR. This problem could arise when a call is connected to a phone or when a call is in a conference on a trunk that does not have line supervision from the CO.

1. As of v5.0, "Ad Hoc" Conferencing refers to the traditional system conferencing feature, whereby an internal user manually adds one or more internal or external parties to a conference call using the phone's conferencing feature.

In a network setting, a user can build a conference with party members (phones and/or trunks) on other nodes. The conference circuits used to build the conference will be allocated from the originating party's node. Any party, not just the originating party, can add a party member to the conference.

- If the party who adds the new member is on the originating party's node, the party will be added to the conference circuit already allocated. This conference is still restricted to the four party limit.

- If the party who adds the new member is not on the originating party's node, a new conference circuit is allocated from the second node. Because this allows more than four parties in a conference, users should be warned that voice levels can be degraded as more parties are added to the conference.

NOTE	If the number of parties in your conference exceeds the capacities of the system resources, you hear reorder tone and the display shows MAX NUMBER OF PARTIES EXCEEDED. To complete the conference setup, return to the parties on conference wait hold one-by-one. Display phone users press the associated CALL or IC button, and single line phone users enter the Individual Hold feature code twice. Then, release parties until there are fewer than the maximum. Then connect the conference.

PROGRAMMING AD HOC CONFERENCING

The available Ad Hoc Conference types are Basic (default) and Advanced. The Basic type supports a maximum Ad Hoc Conference size of four parties. The Advanced type supports a maximum Ad Hoc Conference size of eight parties. When upgrading a system from previous versions of databases, the Basic type is selected by default to retain functionality of previous versions. New installations also default to Basic.

Although the Advanced Ad Hoc Conferencing option provides increased Ad Hoc Conferencing capacity, there are circumstances where it makes sense to use the Basic Ad Hoc Conferencing setting instead. For example, with systems that are already using most or all of their available IP resources, administrators may not want to have any of those resources used for Ad Hoc Conferencing. Refer to "Appendix E: Engineering Guidelines" in the *MiVoice Office 250 Installation Manual* .

Note that if the Advanced option is enabled for Ad Hoc Conferencing and the system is licensed for Meet-Me Conferencing, the two conferencing applications share conferencing resources.

To change the Ad Hoc Conference type:

1. Select System – **Conference-Related Information**.

2. Change the Ad Hoc Conference Type to **Basic** or **Advanced**.

NOTE	Changes to this field require a system reset.

MEET-ME CONFERENCING

This section contains the following information:

- **Feature Description** below

- Programming Meet-Me Conferencing on **page 285**

- Troubleshooting Meet-Me Conferencing on **page 287**

FEATURE DESCRIPTION

The Meet-Me Conferencing feature allows up to eight internal and/or external users to dial into a conference instead of being manually added by an internal user, as is the case when using the system's traditional Ad Hoc Conferencing method.

Many existing system features that work with the traditional Ad Hoc Conferencing method (such as record-a-call, conference transfer, conference hold, etc.) also work with the Meet-Me Conferencing method. Note that unlike an Ad Hoc Conference, a Meet-Me Conference can consist entirely of outside parties; an internal system user does not have to be present.

Release 5.1 and later increased the Meet-Me Conferencing limits from 20 ports with a maximum conference size of eight parties, to 40 ports with a maximum conference size of twenty parties.

Conference Assistant

Conference Assistant is a voice-guided application that allows callers to initiate or join a Meet-Me Conference by entering the proper access code. Callers can access the Conference Assistant by dialing or being transferred to its assigned extension number. The Conference Assistant can also be accessed as a trunk group ring-in destination or through a voice mail automated attendant application.

If users enter an invalid access code, the Conference Assistant prompts them to try again (up to three times). The Conference Assistant also lets users know if there are no conferencing circuits available or if the conference is already at capacity.

The first (and last) participant in the conference hears music. And, if anyone else joins or leaves the conference, the existing participants hear a notification tone to signal the change.

If the conference contains only external callers, then the Conference Assistant will prompt the conference participants to extend the conference after a period time. If no one presses a digit to extend the conference, the system terminates the conference.

The Conference Assistant provides voice-guided prompts for all languages supported by the system. For example, when an internal user calls the Conference Assistant, the application uses the device's assigned language to play the voice-guided prompts. And, when an external user calls in to the Conference Assistant, the application uses the trunk's assigned language to play the voice-guided prompts. Note that the voice-guided prompts cannot be customized.

See "Programming Meet-Me Conferencing" on for information on creating a Conference Assistant and assigning an extension number to it.

Licensing

Meet-Me Conferencing requires a Meet-Me Conferencing software license.

Access Codes

When Meet-Me Conferencing is enabled, users have the ability to initiate or join a Meet-Me Conference by dialing the Conference Assistant and entering a valid access code. Access codes are generated to match each "User" main extension number in the system (as programmed in the Users folder), so that each designated "User" (not phone) has their own access code for establishing conferences.

In addition to the personal access codes that match each "User" main extension (as programmed in the Users folder), the User Web Portal application (see **page 458**) can also be used to create unique system-generated access codes.

Procedure

For specific instructions on using the Meet-Me Conferencing feature, refer to the *MiVoice Office 250 Phone User Guide Supplement for Version 5.0 Software*. These instructions will also be incorporated into the various active phone user guides for the General Availability (GA) software release.

Note that the traditional Ad Hoc Conferencing feature is still available for use even when Meet-Me Conferencing is licensed and enabled. Both conferencing types are available for use as needed. Also, display phones (on nodes running v5.0 or later software) show the following for both conferencing types: the name of the conference [if available], the total conference duration, and the number of parties connected to the conference.

Multi-Node Support

If a Conference Assistant is exported to other nodes in the network, an off-node Conference Assistant (which appears as an off-node phantom) may be called to access a Meet-Me Conference on another node. A ring-in destination may also be an off-node Conference Assistant.

PROGRAMMING MEET-ME CONFERENCING

The following section provides instructions for programming Meet-Me Conferencing.

NOTE	You must have a Meet-Me Conferencing software license uploaded to the system.

To enable Meet-Me Conferencing:

1. Select System – Devices and Feature Codes – **Assistants**.

2. Right-click anywhere in the right pane, and then select **Create Conference Assistant**. The Create Conference Assistant dialog box appears.

3. Select or type the extension in the Starting Extension drop-down list box. You can select one extension.

4. Click **OK**. The extension appears in the Assistants folder.

To configure Meet-Me Conferencing:

1. Select System – **Conference-Related Information**.

2. Select **Meet-Me Conferencing**. Program the following information:

    •   **Conferences**: Displays all user-created conferences on the system. Although each User inherently has a conference access code that matches their main extension number as assigned in the Users folder, these are not displayed in this folder or anywhere else in DB Programming. System administrators cannot create conferences from DB Programming; instead users must use the Configuration Assistant or the User Web Portal. Conferences are displayed with the following information:

        •   **Access Code**: The code entered to join to the conference.

- • **User Name**: The associated user's Last, First name.

- • **Date Last Used**: The date when the conference was last used.

- • **Time Last Used**: The time when the conference was last used.

- • **Unsupervised CO Conference Timer (minutes)**: After one hour, the system will end a Meet-Me Conference containing only outside parties unless a participant dials "1" when prompted to extend the conference. If extended, this prompt will repeat at an interval determined by the Unsupervised CO Conference timer. The range is 1-255 minutes; the default is 15 minutes. Note that parties will not be prompted within the first hour of a conference. If an internal party joins the conference, the timer is canceled.

- • **Access Code Length**: Determines the length of the access code to join the conference. The default length is 7 characters.
  You can also use your extension number as the access code to join a conference.

- • **Audio for First Party**: Enter the file name of the music that the first party hears while waiting for others to join. It is set to default_moh.n64u by default.

To select a file name in online mode:

1. Right-click the Value column in the Audio For First Party field, and then select **Assign File**. The Assign File Based Music-On-Hold dialog box appears.
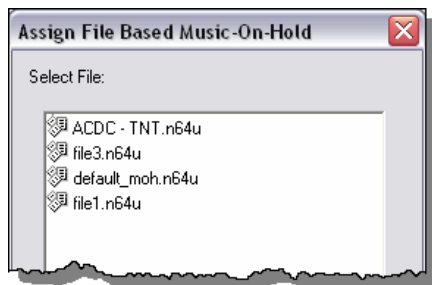
8. Select the file name from the list, and then click **Assign**.

To unassign a file name in online mode:

Right-click the Value column in the Audio For First Party field, and then select **Unassign File**. Any file name that was previously in the field is cleared.

To select a file name in offline mode:

Type the file name you want to use in the Value column in the Audio For First Party field.

NOTE	If there is a file name assigned to this field that does not exist on the system, a warning message appears when you connect to a system using DB Programming. Also, the system does not need to be licensed for File-Based Music-On-Hold to use the MOH files for the conference audio.

To allow Configuration Assistant users to manage their Meet-Me Conference access codes, each phone user's extension must be assigned to the main extension of a User.

To assign the phone extension to the main extension of a User:

1. Select Users – *<User>*.

2. Set the Main Extension to use the extension of the phone.

For Configuration Assistant programming instructions, see "Programming Configuration Assistant" on **page 461**.

TROUBLESHOOTING MEET-ME CONFERENCING

If users do not hear the option to manage Meet-Me Conference access codes when using the Configuration Assistant feature, make sure that each user is programmed with a main extension number in the Users folder within DB Programming.

To display high-level conferencing information and details about each conference using the System Administration & Diagnostics application, refer to the Conferences Content Control help topic in the System Administration & Diagnostics Help.

In a multi-node environment, if a display phone is on a remote node running pre-v5.0 software, it shows the number of parties in the conference. However, if the display phone is on a remote node running v5.0 or later software, it shows the number of parties in the conference.

CONFIGURATION ASSISTANT

Configuration Assistant is a voice guided configuration portal that provides easy-to-use, remote access to the following end-user phone configuration options:

- Dynamic Extension Express

- Do-Not-Disturb (DND)

- Call Forwarding

- Administrator Functions

See "Configuration Assistant" on **page 458** for details.

DEFAULT STATION

A phone user can enter one feature code that will return the following features to their default states. Returning to default:

- Cancels the following features, if in effect:
    - Do-Not-Disturb
    - Manual Call Forwarding
    - Queue Callback Request
    - Account Code For All Calls Following
    - Background Music
    - Ring Intercom Always
    - Headset Mode

- Restores the following features, if disabled:
    - Handsfree
    - Page Receive
    - Hunt Group Replace
    - System Forwarding

- Returns phone volumes to default levels

- Returns phone to standard keymap

- If the phone is a member of one or more ACD hunt groups, logs into the hunt group(s)

This feature is especially useful for installers and troubleshooters who need to know exactly how a given phone is programmed.

For feature usage instructions, refer to the applicable phone user guide.

DIRECTORY OF INTERCOM, SPEED DIAL, AND FEATURE CODES

The Directory feature enables display phone users to look up intercom extension numbers and user names, System Speed Dial numbers and names, or system feature codes. Once the desired information has been displayed, the user may automatically dial the number, store the number in a Station Speed Dial location, select a different number/name, terminate the directory routine, or switch to the other directory. A directory search can be performed when a call is waiting on Conference Hold or Transfer Hold at the phone.

> **NOTE** The Directory feature is only available on Mitel phones, and the standard Directories feature license is required.

In a network setting, a user can see intercom directory names and extension numbers for all devices that are programmed as Off-Node Devices on the local node. Devices that are represented by wildcard off-node extensions do *not* show up in the intercom directory.

For feature usage instructions, refer to the applicable phone user guide.

For programming instructions, see "Programming Device Descriptions and User Names" on **page 379**.

To use the directory, the user enters a letter, a string of letters, a valid extension number, or a valid feature code. If searching for a name, the full name need not be entered. The system will find the closest match and show the number and its associated name on the phone display. Or, the user can scroll alphabetically through the stored list of names. (It is not possible to scroll through the extension numbers or feature codes numerically; extensions and features scroll alphabetically.)

Directory names can include US English, UK English, Canadian French, Japanese, and Mexican Spanish characters, or a combination.

> **NOTE** When using Japanese as the Primary or Secondary language, you can only search for IC and CO directory names based on the last name. You cannot search based on the first name.

Keypad buttons are used to enter the desired letters, numbers, and punctuation. For more information, refer to the applicable phone user guide.

The intercom directory is automatically updated whenever user names and/or extension number information is reprogrammed. The outside directory is updated whenever a System Speed Dial number or name is reprogrammed.

Only Administrator phones will be able to view Administrator feature codes using the directory. If the Diagnostics Mode feature is enabled, the Administrator will also be able to see the diagnostic feature codes.

See "Multilingual Capability" on **page 314** for details about supported languages.

Intelligent Directory Search

The Intelligent Directory Search (IDS) feature simplifies searching for entries in a directory by significantly reducing the number of keystrokes required to find a match. However, because

the dialpad has fewer buttons than the number of letters in the alphabet, each button represents several letters. For example, dialpad button **5** represents "5, J, K, L, j, k, or lower case L." To enter JONES, press **56637**. As you press buttons, several possible matches appear on the display phone screen. This data entry process is similar to the "text on nine keys (T9)" feature found on some cell phones.

With minimum keystrokes, users can retrieve the names and phone numbers of persons entered into the MiVoice Office 250 Intercom (IC) and Outside Directories and identify the status of features on phones from the Features Directory.

For feature usage instructions, refer to the applicable phone user guide.

| NOTE | The IDS feature does not apply to the Voice Mail Directory. Continue to use existing user guide instructions for entering and retrieving names from the Voice Mail Directory. |

A display phone or an 8602 IP softphone application is required for using the IDS feature. The 8602 IP softphone is a software application that enables Voice over Internet Protocol (VoIP) telephone calls from laptop and desktop computers.

DO-NOT-DISTURB

Placing a phone in Do-Not-Disturb (DND) halts all pages, incoming intercom calls, camped-on calls, and transferred calls to that phone. Queue callbacks, recalls, and direct ring-in calls are not blocked. Another user calling the phone while it is in Do-Not-Disturb hears a repeating signal of four fast tones and a pause. Display phones show the Do-Not-Disturb message. The caller cannot camp on, but can queue or leave a message at the phone.

For complete information about DND, see "Do-Not-Disturb" on .

DOOR RELAY CONTROL

The HX Controller chassis includes a relay jack that can be used to unlock a door (that is, "buzz" it open) by dialing a feature code. For details on hooking up a door relay control mechanism to the relay jack on the chassis, refer to the *MiVoice Office 250 Installation Manual* .

NOTE	Door Relay port contacts are rated at 500mA @60 Vac or Vdc peak and are normally open.

To support this feature, there is a phone flag in DB Programming called Activate Door Relay. By default, this flag is disabled for all users. Only phones that have this flag enabled can use the Activate Door Relay feature code (332) to control the door relay. When this feature code is entered at an authorized phone, the door relay jack on the chassis is activated for a period of time defined by the Door Relay Duration timer (default is two seconds), then deactivated automatically.

The Activate Door Relay feature code can be entered while idle or while on a call. This allows users to activate the feature while talking to a communications device mounted at the door without first having to hang up. When activated while on a call, the feature will not affect the call in progress other than to show a short confirmation on the display.

If a user activates the door relay control feature while the relay is already active, the door relay timer is re-posted, effectively causing the door relay to remain active for a longer period of time. For example, assume the timer is set to five seconds. User A enters the feature code, and three seconds later, user B enters the feature code. The relay will deactivate five seconds after user B enters the feature code, thus making it active for a total of eight seconds. This also applies if the same user that initially deployed the feature code does it again. If there is a system reset while the relay is activated, it will be deactivated and remain that way when the system comes back up.

The Activate Door Relay feature may be added to a phone keymap button, just like any other phone-related feature. And, any user can have a button programmed for the feature, but they can only deploy it if enabled in the database.

See the following pages for programming fields:

- Activate Door Relay phone flag in System\Devices and Feature Codes\Phones\<*phone*>\Flags and System\Devices and Feature Codes\Phones\<*phone*>\Flags (see "Phone/Phantom/Hot Desk Profile Flags" on **page 418**)

- Activate Door Relay feature code—default 332 (see "Phone Feature Codes" on **page 769**)

- Door Relay Duration system timer (see "Timers and Limits" on **page 756**)

Refer to the *MiVoice Office 250 DB Programming Help* for instructions about how to program these fields.

FORWARDING – CALL FORWARDING

With Call Forwarding, a phone user can route incoming intercom and outside calls, including direct ring-in calls, to another phone, to a hunt group, or to an outside telephone number, if allowed by toll and trunk restrictions. In a network setting, the network allows the user to forward calls to phones or hunt groups on another node.

The four forwarding options include:

- **Forward All Calls:** All incoming calls are immediately forwarded.

- **Forward If No Answer:** Incoming calls are forwarded if they are not answered before the Forward No Answer timer expires.

- **Forward If Busy:** Incoming calls are immediately forwarded if the phone is busy.

- **Forward If No Answer or Busy:** Incoming calls are forwarded immediately if the phone is busy, or forwarded if calls are not answered before the Forward No Answer timer expires.

For feature usage instructions, refer to the applicable phone user guide.

If forward all calls is enabled, display phones show the call forwarding status and destination until the request is canceled. If one of the conditional forwards is enabled (no answer, busy, or no answer/busy), display phones show the forward status for five seconds and then return to normal. If the phone receiving the forward is a display phone, it shows EXT XXXX FWD FROM EXT XXXX for each forwarded call received.

Phone users can chain Forwards from phone to phone providing the Forwards do not form a loop and the chain does not include more than 10 phones. The conditional forward features (if busy, if no answer, if busy/no answer) may form a loop that the system cannot detect until a call is placed to the forwarding phone. For example, if two phone users forward their calls to each other using the Forward If Busy feature, the system accepts the requests. However, if a call rings in while both phones are busy, the Forwards create an illegal loop. In this case, the call camps on to the called phone and that phone's display shows INVALID FORWARD PATH.

If more than one phone has ring in for a trunk group, direct ring-in calls on that trunk group forward to extension numbers, but not outside numbers or voice mail ports. The display of the phone receiving the forwarded call shows it as a forwarded call, and the CALL button or individual trunk button flashes to show ring in. The individual trunk button also flashes on the phone that is being forwarded. Calls cannot be forwarded to restricted outside telephone numbers or phones in Do-Not-Disturb. If the phone that is programmed to receive your forwarded calls is later placed in DND and you receive a call, you will momentarily see a display showing that the destination phone is in DND; the call will remain at your phone.

NOTE	Direct ring-in calls that are forwarded to a phone in DND ring the DND phone in accordance with the rules of DND.

If your phone is in DND and you have call forwarding programmed, the call is still forwarded, unless you enabled Forward No Answer. With Forward No Answer, intercom callers see the DND display that you programmed, and the call is not forwarded.

If calls are forwarded to a Voice Processing application, and the system is unable to communicate with the Voice Processing PC, the call is not forwarded. It remains at your phone.Call forwarding overrides system call forwarding at the principal phone.

If a phone is programmed for Forward No Answer or Forward No Answer/Busy and receives a transferred call, the Forward No Answer timer is started when the intercom call from the transferring phone begins ringing. The Forward No Answer timer is restarted if the transferring phone completes the transfer before the timer expires. When the timer expires, the transferred call is forwarded.

Agent Help Request calls, queue callbacks, and recalls do not forward, except that a recall at an attendant's phone will forward to another phone.

FWD BUTTON AND MENU BUTTONS

Some phones use a combination of the default FWD button and forwarding menu buttons. When the default FWD button is pressed, the display shows a menu of call forwarding options. The user must then select the desired option by pressing the corresponding button next to the display. If the FWD button is programmed to use any other Forward feature code (355–358), the forwarding options is not displayed when the FWD button is pressed; the phone operates the same as any other phone.

Any user can make any user-programmable button, including the FWD button, a FWD button that uses one of the forwarding feature codes (355–358). For procedures to program user-programmable buttons, see "User-Programmable Feature Buttons" on **page 340**.

A user-programmed FWD button is lit only when the phone is programmed for the call forwarding condition enabled by that button. For example, if a user-programmed FWD button is set to forward calls when the phone is busy, the button will be lit when the Forward If Busy feature is enabled, but not if the Forward If No Answer feature is enabled.

If a user has both a fixed FWD button and a user-programmable FWD button, the fixed button will always light when the phone is forwarded. However, the user-programmable FWD button will be lit only when the forwarding option activated by that button has been selected. For example, if the user has the Forward All Calls feature programmed under a user-programmable button, that button will light if either the fixed or programmable FWD button is used to select that feature. However, if the fixed button or a feature code is used to set the Forward If Busy feature, only the fixed button will light.

FORWARD TO AN OUTSIDE NUMBER

In order to Call Forward a phone to an outside telephone number, a trunk access code must be programmed in front of the destination telephone number. If the phone is called while the selected trunk group is busy, the call is not forwarded. Calls ring at the forwarded phone until the call is answered or the caller hangs up. The forwarded phone's display shows FORWARD TO BUSY TRUNK.

The forwarded phone's (not the intercom caller's) trunk and toll restrictions are checked when an intercom call is forwarded to an outside number.

When an outside call is forwarded to an outside number, the Unsupervised CO timer is activated. When the timer expires, the call recalls the attendant. If the attendant does not answer the recall before the Abandoned Call timer expires, the call is disconnected.

Phones with Forced Local Toll Call and Forced Long-Distance Toll Call account codes cannot forward calls to outside numbers.

If a trunk group is assigned direct ring in to multiple phones, and one of those phones is forwarded to an outside number, incoming calls on that trunk group are not forwarded to the outside number.

> **NOTE** Some reduction in voice volume may occur when an outside call is forwarded to an outside telephone number, depending on central office trunk quality.

MANUAL FORWARDING TO PUBLIC NETWORK

There is a phone flag called Manual Forwarding to Public Network. When the Manual Forwarding to Public Network phone flag is disabled, it prevents the phone from manually forwarding calls to the public network. This prevents users from calling in to a local phone and reaching toll numbers via forwarding. This phone flag is located in System\Devices and Feature Codes\Phones\Local\<*phone*>\Flags.

FORWARD TO THE MESSAGE CENTER

Phone users can forward calls to their message centers. Users can forward calls to their designated message center by pressing the FWD button and then the MSG button or MESSAGE CENTER menu button. Single-line users must use the call forwarding procedure and dial the extension number of the message center.

FORWARD TO AN ATTENDANT

Phone users can forward calls to their attendant by pressing the FWD button and then 0, or by entering a Call Forwarding feature code and dialing 0.

FORWARD TO VOICE MAIL

If a phone is forwarded to voice mail or if the message center is voice mail, the user's mailbox number is automatically dialed when the voice mail unit answers the forwarded call.

If a chain of forwarded phones ends in voice mail, the mailbox number of the first phone in the chain will be selected when the voice mail unit answers the call.

If a trunk group is assigned direct ring in to multiple phones, and one of those phones is forwarded to a voice mail unit, incoming calls on that trunk group are not forwarded to the voice mail unit.

FORWARD TO E-MAIL

The Forward to E-mail feature enables Unified Voice Messaging (UVM) to forward voice mail messages as **.wav** file attachments to e-mail messages.

Features provide the following benefits:

- Allows each mailbox to be configured with a different E-Mail address.

- Operates with an industry standard SMTP mail server.

- Each mailbox can be designated to store, forward, or store-and-forward messages.

NOTE	If both Remote Messaging and Forward to E-Mail features are enabled, the Forward and Copy option must be selected. If the Forward Only option is selected, the user will not receive Remote Messaging notifications.

Once E-mail Gateway information has been configured in DB Programming, voice mail updates the configuration for E-mail SMTP delivery software to use this information. Voice Mail then enables the E-mail Gateway as active and the user's voice mails may then be forwarded. If this information is not configured, voice mail messages will *not* be forwarded.

When the E-mail Gateway is marked as active, then once the user configures an e-mail address and marks their mailbox as "not disabled" for the e-mail gateway feature, their voice mails will be forwarded to that configured address. The E-mail task starts up and cycles through the global list of mailboxes looking at messages. Any that are marked with e-mail enabled and configured with an e-mail forwarding address, will have all their messages put on the e-mail queue to be sent out to their configured e-mail address. If there were messages left prior to configuring the mailbox with an e-mail address and enabling the e-mail forwarding feature, these messages are not sent. Only those left after enabling the e-mail forwarding feature will be sent, as those messages will be marked accordingly.

FORWARDING – SYSTEM FORWARDING

System Forwarding provides the ability to program the system so that calls ringing at a phone follow a programmed "forwarding path" that routes the call based on the type of call and the status of the intended phone. The Call Forwarding phone feature can be used to override System Forwarding.

For feature usage instructions, refer to the applicable phone user guide.

The following terms describe the System Forwarding feature:

- **Principal Phone**: Original phone being called.

- **Forwarding Point**: A destination (phone, hunt group, or off-node device) to which the system-forwarded call is directed. Each forwarding path can contain up to four forwarding points.

- **Forwarding Path**: The sequence of forwarding points to which the call is sent. Forwarding paths are programmed in the database. Up to 200 unique paths can be programmed. Each phone can have up to three forwarding paths. See "System Forwarding Paths" on **page 469** for details.

Two system timers are used with this feature (see "Timers and Limits" on **page 756** for details):

- **System Forward Initiate:** Determines how long an unanswered call will ring at the principal phone before moving to the first forwarding point. The default value is 15 seconds and the range is 2–255 seconds.

- **System Forward Advance:** Determines how long an unanswered call will ring before moving to the next forwarding point. The default value is 15 seconds and the range is 2–255 seconds.

For each forwarding path assigned to the phone, the system checks the following three criteria to determine if and when a call should be forwarded:

- **Type of incoming call**: Up to six different types of calls can be programmed to be sent to the forwarding path, including:
  - Outside calls received through a call routing table (including DID and E&M calls, but not including DISA calls)
  - Ringing outside calls
  - Transferred outside calls (including automated attendant and voice mail transfers)
  - Recalling outside calls
  - DISA calls (including DISA calls received through a call routing table)
  - Intercom calls

- **Phone status**: The system recognizes four different types of phone status.
  - *No Answer:* If the call is not answered at the principal phone before the System Forwarding Initiate timer expires, the system sends the call to the forwarding path.
  - *Busy*: If the principal phone is busy, the system immediately sends the call to the forwarding path. Both "No Answer" and "Busy" can be selected together to form a "Not Available" status.

- • *Do-Not-Disturb*: If the principal phone is in Do-Not-Disturb, the system immediately sends the call to the forwarding path.

  - • *Immediate*: The system immediately sends the call to the forwarding path. The principal phone user cannot answer an immediately forwarded call, but can reverse transfer it from the forwarding point.

- • **Day-mode/night-mode status**: The system day/night mode is used to determine if a forwarding path is taken. Either day mode, night mode, or both may be specified.

For an example, see the diagram below. In the example, the "principal" phone user wants direct ring-in or transferred calls to ring at his or her phone during day mode. The principal phone has a forwarding path with two forwarding points: a hunt group and voice mail. If the principal does not answer the call, it follows the forwarding path to forwarding point #1, a hunt group. If the hunt group does not answer the call, it would continue on the forwarding path and go to the principal phone's voice mailbox (forwarding point #2).

If the principal phone is a phone, the "ring principal once" option can be set that will signal the principal phone when a call begins to follow the forwarding path. The signal to the principal phone consists of a display (CALL SENT TO FORWARD PATH) and a single burst of ring tone. The call cannot be answered at the principal phone, but can be reverse transferred from the system forward point.

CALL RINGS IN TO PRINCIPAL phone

No Answer

↓

CALL IS SENT TO FWD POINT #1 - HUNT GROUP

↓

CALL IS SENT TO FWD POINT #2 - VOICE MAIL
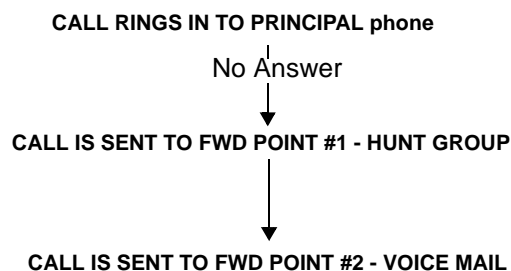
If a call rings in to multiple phones, and one or more of those phones has system forwarding, the call will not follow any of the forwarding paths.

Agent Help requests will not follow any forwarding path.

If a principal phone or a phone forwarding point is a member of a hunt group, calls placed to the hunt group's pilot number are unaffected by system forwarding. The hunt group calls will be received at the phone as usual and will not enter the system forwarding path.

A call follows only the forwarding path of the principal, even if a forwarding point has a forwarding path of its own. The call that originated at the principal phone will follow only the principal phone's forwarding path.

INDIVIDUAL PHONE FORWARDING POINTS

The ringing duration at each phone forwarding point is determined by the System Forwarding Advance timer (defaults to 15 seconds). If the phone forwarding point is busy, the call will camp-on until the System Forwarding Advance timer expires and then it moves to the next forwarding point.

If the phone forwarding point is in Do-Not-Disturb, the forwarding path will bypass that phone and immediately send the call to the next forwarding point.

If a forwarding point is a Voice Processing application, and the system is unable to communicate with the Voice Processor, the call will bypass the forwarding point.

A phone forwarding point can place calls or transfer calls to the principal.

HUNT GROUP FORWARDING POINTS

If a forwarding point is a hunt group, a call following the forwarding path will be sent to the announcement or overflow station as usual; it will advance to the next forwarding point if not answered before the System Forwarding Advance timer expires.

The call will ring at the hunt group until the System Forwarding Advance timer expires. It will then move to the next forwarding point. The No Answer Advance timer determines how long the call will ring at each hunt group phone, as usual.

If all phones in a hunt group forwarding point have Do-Not-Disturb or hunt group remove enabled, the call will camp on until the System Forwarding Advance timer expires. The call will then be sent to the next forwarding point.

HANDSFREE ANNOUNCE SYSTEM FORWARD OPTION

When a call is transferred to the principal phone by another phone user *and the principal phone's system forward is set for the "no answer" phone status*, a system-wide option determines what the caller hears, as shown in the following table.

Table 49:   Handsfree Announce Options

HANDSFREE ANNOUNCE OPTION	SYSTEM FORWARD OPTION	RESULT
Disabled	Intercom and/or outside calls forwarded	The intercom call from the transferring phone enters the forwarding path after the System Forwarding Initiate timer expires. The transferring phone user hears ringing and can announce the call only if it is answered.
Enabled	Outside calls forwarded	The transferring phone user will place a handsfree call to the principal phone and can announce the transfer; the *System Forwarding Initiate* timer does not start until the transfer is completed.
Enabled	Intercom calls forwarded	The handsfree announce option is overridden and the intercom call from the transferring phone enters the forwarding path after the System Forwarding Initiate timer expires. The transferring phone user hears ringing and can announce the call only if it is answered.
Enabled or Disabled	Calls forwarded to voice mail	The transfer is completed when the transferring phone hangs up. The principal phone's mailbox number is automatically dialed.

MANUAL CALL FORWARDING AND SYSTEM FORWARDING

If the principal phone has manual call forward enabled (using the FWD button or one of the Call Forwarding feature codes), the Call Forward feature overrides System Forwarding.
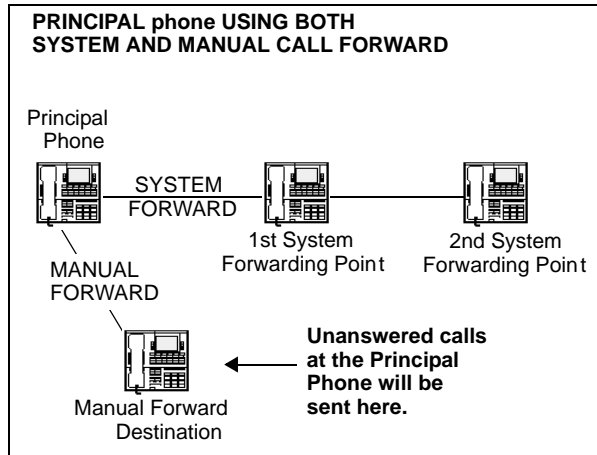
If a forwarding point has the Call Forward feature enabled, a system-forwarded call will ring at the forward destination until it is answered or the System Forward Advance timer expires; then the call moves on to the next forwarding point.
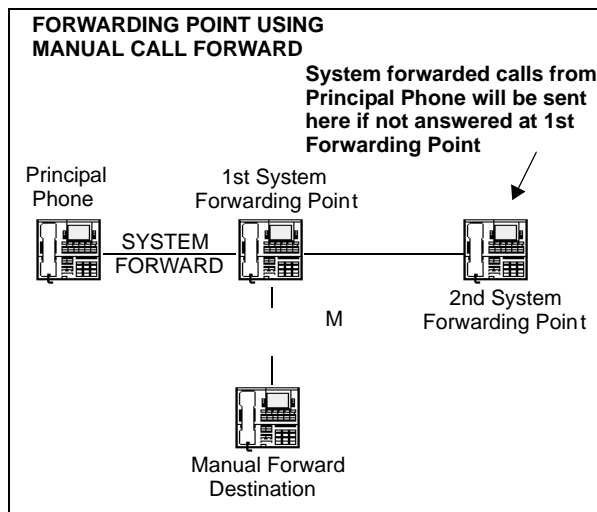
If a phone forwarding point is manually forwarded to voice mail, the call will not follow the manual forward to Voice Processing; it will ring at the phone forwarding point.

FORWARDING POINT
FORWARDED TO VOICE PROCESSING
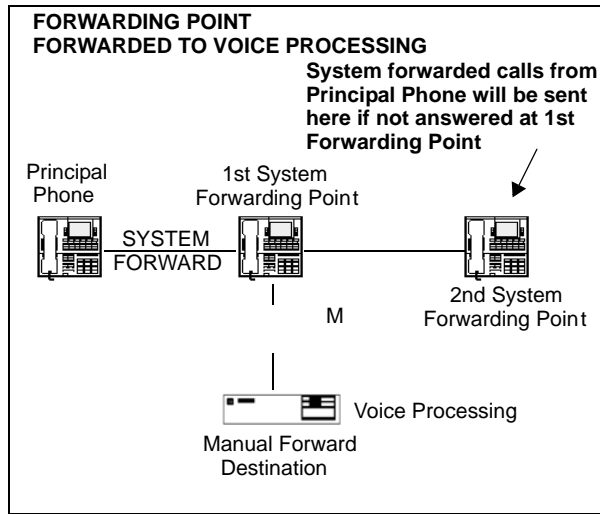
If the principal phone *receives* a manually forwarded call (not a system forward), that call will *not* follow the principal phone's system forwarding path.

PRINCIPAL phone IS MANUAL
FORWARD DESTINATION
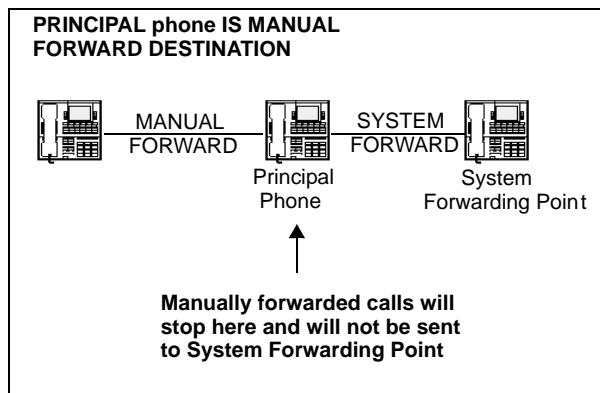
If an infinite forward loop results from the combination of manual forwards and system forwarding paths, the phone that was originally intended to receive the call will ring, even if the phone is in Do-Not-Disturb.

INFINITE LOOP CAUSED BY
SYSTEM AND MANUAL CALL FORWARD
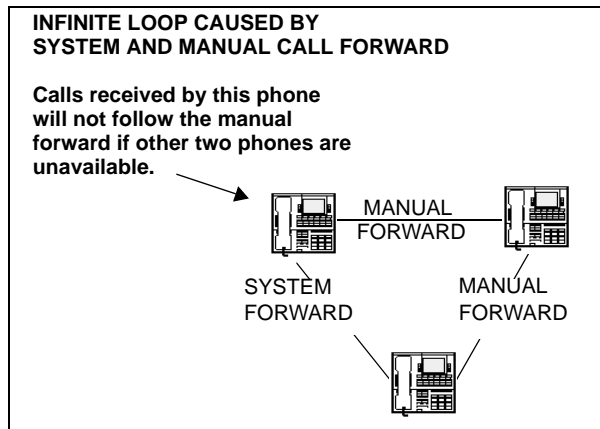
SYSTEM FORWARD ENABLE/DISABLE

The phone user has the option of entering feature codes that will disable or enable the use of System Forwarding at that phone. The enable/disable flag does not affect the phone's appearance on any other phone's call forwarding path. The flag merely determines if the phone will have its own calls System-Forwarded.

UNANSWERED SYSTEM FORWARD CALLS

If a call that is following a system forwarding path is not answered or all forwarding points are in Do-Not-Disturb or busy, its final destination depends on the type of call as shown below.

Table 50:   Destinations of Unanswered System-Forwarded Calls

TYPE OF CALL	ALL FWD POINTS ARE BUSY OR DO NOT ANSWER	ALL FWD POINTS ARE IN DO-NOT-DISTURB
Intercom	Rings at the last forwarding point until it is answered or the caller hangs up.	Call never leaves the principal phone.
Call routing ring-in (except DISA)	Rings at last forwarding point until it is answered or caller hangs up.	Rings at the principal phone until it is answered or the caller hangs up.
Direct ring-in	Rings at last forwarding point until it is answered or caller hangs up.	Rings at the principal phone until it is answered or the caller hangs up.
Transfer from a phone, automated attendant or voice mail	Recalls the transferring phone's transfer recall destination when the System Forwarding Advance timer expires at the last forwarding point.	*If immediate forwarding is enabled,* rings the principal phone until the applicable Transfer timer expires. Then it recalls the transferring phone's transfer recall destination. *If a conditional forwarding is enabled,* rings principal phone until the System Forward Initiate timer expires. Then it recalls the transferring phone's transfer recall destination.
DISA (including Call Routing to DISA)	Rings at last forwarding point until the Abandoned Call timer expires.	Rings at the principal phone until the Abandoned Call timer expires.
Recall	Recalls the principal phone's attendant when the System Forwarding Advance timer expires at the last forwarding point. If the principal phone does not have an attendant, the call continues to ring at the last forwarding point until the Abandoned Call timer expires.	*If immediate forwarding is enabled,* rings principal phone until the Recall timer expires. Then it recalls the principal phone's attendant. *If a conditional forwarding is enabled,* rings principal phone until the System Forward Initiate timer expires. Then it recalls the principal phone's attendant. If the principal phone does not have an attendant, the call continues to ring at the principal phone until the Abandoned Call timer expires.

GROUP LISTEN

The Group Listen feature (feature code 312) allows a user to transmit a conversation over the phone speaker while in handset or headset mode. This allows other people in the room to listen to the conversation. However, the phone microphone remains disabled so that only the headset or handset user can speak.

This feature cannot be used on a handsfree call. The user must be on a call using the handset or a headset before entering the feature code. Group Listen cannot be used on single line sets.

NOTE	When using Group Listen on an 8500, 8520, or 8560 digital telephone, the volume control adjusts the volume level of the headset speaker or handset speaker, not the external speaker.

When the Group Listen feature is active in handset mode, the Speaker lamp remains unlit. This allows the user to place the call into handsfree mode at any time during the call by pressing the Speaker button. When the feature is active in headset mode, the Speaker lamp is lit. Pressing the button disconnects the call.

For feature usage instructions, refer to the applicable phone user guide.

HOLD

The system provides three ways for placing intercom and outside calls on hold. While on hold, the caller hears music if the system is equipped for MOH. The three hold applications include:

- **Individual Hold (feature code 336):** Places the call on hold at one phone. It can then be picked up directly at that phone or it can be picked up at another phone using the Reverse Transfer feature (see **page 339**).

- **System Hold (feature code 335):** Places the call on hold so that it can then be picked up directly at any phone that has an individual trunk button and has allowed-answer and/or outgoing access for the associated trunk, or at the phone that placed it on hold. Attempting to place a conference on system hold will place the conference on individual hold. Intercom calls cannot be placed on system hold. Single line phones cannot place calls on System Hold; attempting to do so at a single line phone will place the call on Individual Hold. In a network setting, when a user puts a trunk on System Hold, only that user and users on the same node as the trunk can access the call.

- **Consultation Hold:** Allows a single line phone user to pause during a call, use other system features, and then return to the caller by hookflashing. If a single line phone user attempts to hang up after placing a call on consultation hold, the call recalls the phone.

For feature usage instructions, refer to the applicable phone user guide.

If a call remains on hold until the Hold timer expires, it recalls the phone where it is on hold, and the Recall timer is started. If it is still unanswered when the Recall timer expires, it recalls the phone's attendant, and the Abandoned Call timer is started. If the phone does not have an attendant, the call continues to recall at the phone that placed it on hold. If the call is not answered before the Abandoned Call timer expires, the call is disconnected by the system.

For users' convenience, the system has two Hold timers: *Hold* and *Hold - Alternate*. In the default state, the Alternate timer is set for a longer time period than the Hold timer. However, both timers are programmable. The "Alternate Hold Timer" phone flag determines which timer each phone will use. If the flag is disabled, the phone uses the Hold timer. If it is enabled, the phone uses the Alternate timer.

NOTE	Phone users can avoid the Hold timer by muting the microphone during a call instead of placing the call on hold. If this is done, the caller will not hear music-on-hold and will not recall the phone.

HOOKFLASH [RECALL]

A timed hookflash [recall, in Europe] may be required for phone and single line phone users to use certain telephone company or PBX features. The CO Hookflash feature code sends a timed hookflash/recall over the trunk when entered. A hookflash/recall restarts the call cost display and toll restriction, plus it starts a new line in the SMDR printout. However, the call remains under the same CALL button.

The Hookflash feature can be enabled or disabled on a trunk group-by-trunk group basis. It can be used on any outside call, including conference calls.

The duration of the hookflash/recall can be changed by reprogramming the LS/GS CO Hookflash timer (default value is 0.6 sec). If the system is not installed behind a PBX, the Hookflash timer must be programmed to be compatible with system trunks.

For feature usage instructions, refer to the applicable phone user guide.

HUNT GROUPS

The Hunt Group feature permits calls to be placed to a group of phones and to be automatically transferred to an available phone in the group. You can program up to 300 hunt groups. Hunt group lists can contain individual phones or extension lists. Non-ACD hunt group phones must reside on the same node; off-node devices must be ACD hunt group members.

There are two types of hunt groups:

- **UCD Hunt Groups**: Included with the system license. UCD hunt group members are automatically logged in to the hunt group. UCD hunt group members cannot log in or log out of hunt groups.

- **ACD Hunt Groups**: Requires a software license. ACD hunt group members log in to the hunt group to receive calls.

For complete information about Hunt Groups, see "Hunt Groups" on **page 552**.

INTERCOM CALLS

The intercom can be used to place phone-to-phone calls that can be answered handsfree. Or, it can be used to place private, non-handsfree, calls. A phone user that reaches a busy phone can camp on, request a callback queue, or leave a message. Additional features that apply to both outside and intercom calls include Call Waiting, Call Transfer, Reverse Transfer, Call Forwarding, and Hold.

> **NOTICE**
>
> When the procedures tell you to hookflash [recall], quickly press and release the hookswitch. If you press the hookswitch to hang up, hold it down until the SL Hookflash Maximum timer expires (default value is 1.2 seconds); otherwise, the system recognizes it as a hookflash [recall].

The network allows handsfree intercom calls when calling from one node to another, unless the call camps on to the node before being sent to the other node.

PROGRAMMING FOR PRIVATE INTERCOM CALLS

A phone user can place private, or non-handsfree, intercom calls by pressing the pound (#) button before dialing the extension number. Or, the phone can be programmed to always send and/or receive private calls using the Ring Intercom Always or Handsfree Disable features.

RING INTERCOM ALWAYS

A phone user can always place private calls by programming the phone with the Ring Intercom Always feature code (377). While this feature is enabled, the called party hears repeating double tones and in order to answer must lift the handset or press the Speaker button, ANSWER button, or IC button. If the phone does not have an IC button, the call appears on a CALL button. In the default database, all single line phones have this feature enabled.

For feature usage instructions, refer to the applicable phone user guide.

PHONE HANDSFREE ENABLE/DISABLE

A user can always receive private calls by disabling the phone Handsfree Answering feature using the Handsfree On/Off feature code (319), as described in the following paragraph. The user hears repeating double tones when receiving an intercom call and must lift the handset (or press the SPKR, ANSWER, or IC button) to answer while Handsfree Mode is disabled.

For feature usage instructions, refer to the applicable phone user guide.

RECEIVING INTERCOM CALLS

If the phone does not have an **IC** button, the call appears under a CALL button. For feature usage instructions, refer to the applicable phone user guide.

PLACING INTERCOM CALLS

When placing an intercom call, begin dialing before the Dial Initiation timer expires. If the timer expires, the system drops the connection and sends reorder tones. And, if you want to place

a private (non-handsfree) call to a phone, press the pound (**#**) button before dialing the extension number.

If you need to look up a number in the IC Director*y*, see the detailed instructions in "Finding an Entry in the IC Directory" on **page 312**.

For feature usage instructions, refer to the applicable phone user guide.

INTERCOM CAMP ON AND QUEUE CALLBACK

Camp On and Queue Callback requests allow the caller to wait for an available resource. A phone user waiting for a specific resource (a trunk or a phone) will be served before a phone user waiting for a group (trunk group or hunt group) that contains that specific resource. This includes camped on Emergency Call feature calls—if the Emergency Call feature uses a trunk group, requests for specific trunks are processed ahead of the Emergency Call. For details about the Emergency Call feature, see "Emergency Calls" on **page 182**.

CAMP ON

When a phone user calls a busy phone or hunt group, the system sends a busy signal. The caller can wait off-hook to camp on (after the Camp On timer expires) and hear music while waiting until the called phone is available. The system periodically sends call waiting signals to the busy phone(s). A user can camp on to busy phones on other nodes and will hear the other node's Music-On-Hold while camped on.

If a phone activates DND while an intercom call is camped on, the camped-on caller is removed from the camped on state and that caller receives DND indications. Intercom callers cannot camp on to a phone that is in DND.

For feature usage instructions, refer to the applicable phone user guide.

BUSY PHONE QUEUE CALLBACK

When a called phone rings busy or is in DND, the caller can request to be queued up for a callback when the phone is available. The calling party activates the feature by entering a feature code or by pressing a menu button and then hanging up This feature can be activated even if the call camps on.

The following are Queue Callback feature usage conditions:

• Each phone can have only one active queue callback request at a time.

• The Queue Callback feature can also be used when calling a hunt group if all phones are unavailable.

• Users can queue onto busy phones and trunks on other nodes.

• Queue callbacks must be answered before the Queue Callback timer expires. If a callback is not answered, the queue is canceled. If the phone is busy when called back, the queue request is placed behind any other waiting queue requests.

• Queue callbacks do not follow forwarding and are not blocked by DND mode.

For feature usage instructions, refer to the applicable phone user guide.

MESSAGES

Intercom callers may leave a message waiting indication if a called phone is busy, if there is no answer, if the phone is in DND, or when they are connected to or placed on hold by another phone. Users can leave a message for and respond to messages from, users on other nodes. There are two messaging options:

•   **Have the called party return your call**: Display phones show the source and time of the message. When the called party responds to the message indication, a call is automatically placed to your phone.

•   **Leave a message with the called party's message center**: Display phones show that a message was left with the user's assigned message center. When the called party responds to the message indication, a call automatically connects to the appropriate message center.

There are feature codes associated with messages. For information about the Message feature codes, see **page 769**.

For feature usage instructions, refer to the applicable phone user guide.

To signal that a message is waiting, a called phone's Message button and Message lamp (if applicable) flash and the display shows the number of waiting messages. Each time the Message button is pressed, the display shows the message source for each of the waiting messages in the order they were received.

For messages from phones, the display shows MSG: *<phone>* and the date and time of the message. For messages from voice mailboxes, the display shows XX MESSAGES FROM MBOX XXXX to indicate the number of waiting messages in each mailbox that left a message. (This is especially helpful at phones with multiple mailboxes.) For single line phones, a system programming option can be enabled that sends six short Message Waiting tones when the user lifts the handset or presses the hookswitch.

NOTE	The Message Waiting Indicator, which is available on IP phones, is programmable. If the MSG button flashes but the Message Lamp does not, the Message Lamp may have been reprogrammed for another feature. Ask your System Administrator for programming details.

Any phone, Voice Processing application, hunt group, or off-node device can be designated as the message center for a phone. However, a phone cannot be programmed as its own message center.

If the designated message center is a voice mail hunt group, the voice mail hunt group is called after the Message Wait timer expires. When the voice mail unit answers the call, the called party's "mailbox" is automatically dialed. The caller can then leave a message in the mailbox. The called party's Message button flashes, and the message display indicates that the message was left by the voice mail hunt group.

Each phone user can leave message waiting indications at more than one phone. If more than one message is received from one phone, the message display will show only the first message, and all other requests will be ignored.

When a phone is forwarded or has system forwarding, and a caller leaves a message waiting indication after calling the forwarded phone, the message indication appears at the original phone instead of the phone that received the forwarded call.

ALTERNATE MESSAGE SOURCE

The Alternate Message Source feature enables a voice mail unit to leave messages through a phone or hunt group phone that is designated as its alternate message source. When a message is left on the voice mail unit for a phone user that has an assigned mailbox, the called phone receives the Message Waiting indication. If it is a display phone, the display shows MESSAGE RECEIVED FROM <*alternate message source*> rather than from the voice mail circuit. When the phone user retrieves the message, an intercom call is placed to the alternate message source.

To make efficient use of multi-port ***analog*** voice mail units the ports are placed in a voice mail hunt group and the voice mail hunt group's pilot number is assigned as the alternate message source for each of the individual voice mail ports. When a user responds to a message left by one of the voice mail ports, the pilot number is automatically dialed, and the call circulates through the hunt group until a voice mail port is available. Without the alternate message source hunt group, the call would return only to the port that left the message and would not circulate through the hunt group.

A phone's message center or alternate message source does not need to be on the same node as the phone. It can be a phone, hunt group, Voice Processing application, or any off-node device.

SILENT MESSAGES

Silent messages can be placed while on-hook or off-hook without making an intercom call to the phone. This method should be used by analog voice mail units.

For feature usage instructions, refer to the applicable phone user guide.

MICROPHONE MUTE

Whether handsfree or using the handset, you can temporarily turn off your microphone while on a call. The call is still connected; you can hear the other party, but they cannot hear you. Since the call is not placed on Hold, no timer is activated. The MUTE button is lit when the microphone is muted; the light goes out when you press the MUTE button to re-enable the microphone.

You cannot mute the microphone on your phone while your call is on Hold at another phone.

For feature usage instructions, refer to the applicable phone user guide.

FINDING AN ENTRY IN THE IC DIRECTORY

The following sections show examples of display phone screens. You will see similar examples when using the IDS feature to retrieve an entry from the IC Directory. Also discussed are the actions you need to take when using the feature.

NOTE	Two-line display phones show only the first two lines of the six lines shown in the following examples. To access the Directory feature on a two-line display phone, press the Special button followed by 307.

If you have not entered digits in the IC Directory, you see the first lines of the entire directory. Empty matches are displayed first. Instead of displaying a blank line, IDS shows the extension in brackets for any match that is empty. The following example shows that the first three IC Directory extensions of the system are not assigned in the database.

The underscore character indicates the number of characters entered. In this example, only the underscore character is displayed because no digits have been entered.

MENU BUTTONS   **MENU BUTTONS**

```
_
11498
► <11498>    ◄
► <11499>    ◄
► <11714>    ◄
►    1/16    >>  ◄
```

ARROW MENU BUTTON
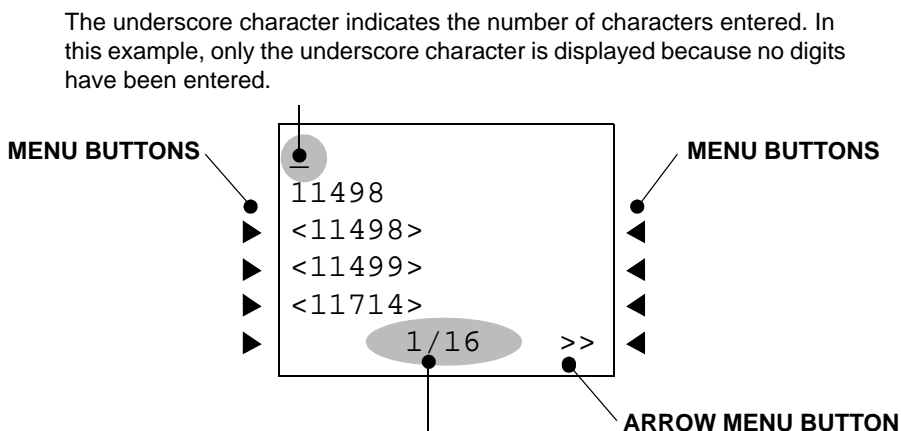
The bottom line displays two numbers separated by a forward slash. The first number indicates which entry in the list of matches is the current entry, and the second number is the total number of matches (exact and partial) for the current input. For example, "1/16" indicates that the first entry refers to the first match out of 16. If the screen cannot display this information, IDS displays no information. For example, "10000/15200" will not fit on the display between the arrow menu buttons (**<<** and **>>**).

FINDING LISTINGS IN THE OUTSIDE DIRECTORY

Upon accessing the Outside Directory, you see the first entries the entire directory. If the directory is empty, a message appears indicating that entries are not available, and you hear

a fast busy signal. If desired, phone numbers can be programmed as "Private" in the database and will appear as a "PRIVATE NUMBER" in the display, as shown in the following example.

UNDERSCORE CHARACTER

MENU BUTTONS   **MENU BUTTONS**

```
EEE_ STORE
PRIVATE NUMBER
EEE STORE
EFG SYSTEM
EGH STORE
<<   21/35      >>
```

ARROW MENU BUTTON   **ARROW MENU BUTTON**

TOTAL NUMBER OF MATCHES
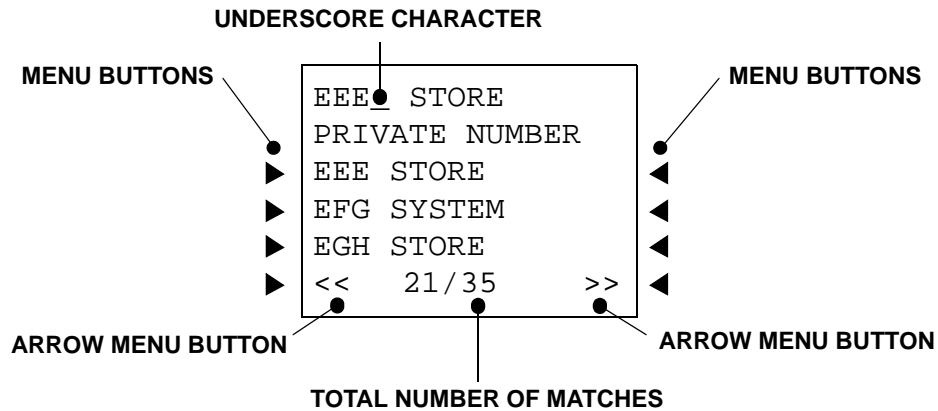
FINDING FEATURES IN THE FEATURE DIRECTORY

The following examples depict Feature Directory displays. Upon accessing the Feature Directory, you see the first entries of the entire directory. Like the IC and Outside Directories, a six-line display phone shows three possible matches at a time. The entries on the first and third lines refer to the same extension. The menu buttons, arrow menu buttons, the total number of matches, underscore character, and dialpad buttons in the Feature Directory function the same as in the IC and Outside Directories.

UNDERSCORE CHARACTER

MENU BUTTONS   **MENU BUTTONS**

```
_ACCT CODE FOLLO
391
ACCT CODE FOLLOW
ACD AGENT LOGIN
ACD AGENT LOGOUT
      1/81      >>
```

ARROW MENU BUTTONS

TOTAL NUMBER OF MATCHES
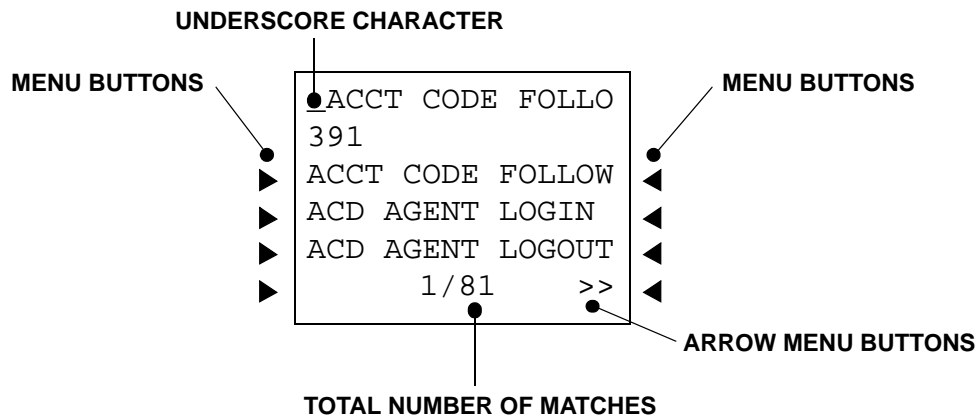
MULTILINGUAL CAPABILITY

This section contains the following information:

- **Feature Description** below

- Phones on **page 314**

- Trunk Language on **page 319**

- Multilingual Do-Not-Disturb and Reminder Messages on **page 320**

- Voice Processing on **page 320**

- Multilingual Feature in Network Operation on **page 321**

- Using Multilingual Directories on **page 321**

FEATURE DESCRIPTION

The MiVoice Office 250 provides a choice among American English, British English, Canadian French, Japanese, and Spanish prompts and displays. The system selects the language to use for each call, as determined by the trunk, phone, and Voice Processing programming, as described in the following sections. Among the IP phones, only the 8660 displays Japanese prompts.

NOTE	52xx/53xx-series IP phones do not support the Japanese language.

Any of these languages can be designated as primary or secondary languages in the system. For information about programming the primary and secondary languages for phones, see "Languages" on **page 406**.

The Change Language feature code (default is 301) allows phone users to switch between the system's designated primary language and the phone's designated secondary language. Once a language is selected, all of the displays on that phone will represent the chosen language (except for diagnostic displays, which are presented in English only, and custom feature and trunk labels, which are presented as programmed). Likewise, all of the voice prompts on that phone will represent the chosen language.

NOTE	If any phones in the system are using Canadian French, all 5000 nodes must be upgraded to v4.0 or later to avoid cross-node display issues.

PHONES

The following subsections provide information on phone multilingual capabilities.

LANGUAGE SELECTION

The platform can be programmed to use a primary language and a secondary language. The available languages are American English, British English, Canadian French, Japanese, and Spanish. A phone flag in database programming determines the language that will be used by each phone.

For example, if the phone is programmed for American English, all phone displays will appear in American English. Also, when the phone is used for calling a Voice Processing application, the voice prompts will be in American English, unless overridden by a Call Routing Announcement. If programmed for Japanese, the displays will appear in Japanese (Katakana) characters and the voice prompts will be spoken in Japanese, unless overridden by a Call Routing Announcement. By default, all phones are set for the system's Primary Language.

All displays and default messages in the system are provided in both languages. Phone-programmed messages, including those programmed through the Administrator's phone, can include specific language characters, or a combination.

When programming a feature that requires a time and date, a user of a phone with English or Spanish as the primary language will enter the date in the order "month, day, year" and the time in the order "time, AM/PM." If Japanese is the primary language, the user enters the date as "year, month, day" and the time as "AM/PM, time." The displays show the time and date in the same order as programmed. For example, the English or Spanish display shows the time and date as "12:25 TUE NOV 28" and the Japanese display shows "TUE 11/28 12:25."

SECONDARY LANGUAGE SELECTION

Each phone in the system has a programming field labelled "Secondary Language." This field corresponds to the Change Language feature (301), which is used to toggle between the system primary language and secondary language. With this feature, the user can toggle between the system primary and secondary language, or can specify a different secondary language. This flexibility at the phone allows the system to support more than two languages.

If a phone's secondary language field is programmed with Use Primary Language, the Change Language feature will do nothing because the phone will toggle between the system primary language and the phone's secondary language, which is the system primary language.

If a Secondary Language field is programmed with Use Secondary Language, the Change Language feature toggles between the system primary language and the phone's secondary language, which is the system secondary language. This arrangement is the system default.

If a phone's secondary language field is programmed with Japanese, the Change Language feature will toggle between the system primary language and the phone's secondary language, which is Japanese.

The Language field for phones indicates what language the phone is currently set to. It can be set to any specific language along with the Use Primary Language and Use Secondary Language options.

An undesirable side-effect of changing the Language field is that if you change it to Japanese and the phone's secondary language field is set to Spanish, then the user will have no way to get back to Japanese if they enter the Change Language feature code. This is because the first time the user enters the feature code, the system toggles the phone to the Use Primary Language. The next time the user enters the feature code, the system toggles the language to the phone's secondary language, which is Spanish. To avoid this situation, simply change the phone's secondary language to Japanese and the phone's Language field to Japanese.

CUSTOM CHARACTERS

Various features, such as Do-Not-Disturb, Station Speed Dial, and so forth, allow users to enter custom characters via their phone's dialpad. Depending on which language the phone is using, the custom characters may change. The charts on the following pages show which custom characters the user will enter depending on the number of times each dialpad button is pressed. Users may enter both lowercase and uppercase characters instead of the uppercase-only restriction in place prior to v4.0.

The following table summarizes which phones use which character bitmaps when the system is equipped with v4.0 software. See the following pages for character bitmap charts.

Table 51: Phones and Supported Character Bitmaps with Version 4.0 Software

	NEW CHARACTER BITMAPS			OLD CHARACTER BITMAPS		
PHONES	US & UK	SPANISH	FRENCH CANADIAN	US, UK, FR-CAN	SPANISH	JAPANESE
52xx/53xx	✓	✓	✓			
8560/8660	✓	✓	✓			✓
8528/8568	✓	✓	✓			✓
All Others				✓	✓	✓

Phones Supporting New Character Bitmaps

The Mitel 52xx/53xx IP phones, the 8660 IP phone, and the 8528, 8568, and 8560 digital telephones are able to support all of the new character bitmaps required to fully support the Canadian French and Mexican Spanish languages. When using these phones, the following charts show which characters the user will enter depending on the number of times each dialpad button is pressed.

Table 52: Custom Dialpad Characters — US English and UK English

BUTTON	NUMBER OF TIMES DIALPAD BUTTON IS PRESSED														
	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
1	:	-	/	,	.	;	(	)	&	+	*	!	?	#	1
2	A	B	C	2	a	b	c								
3	D	E	F	3	d	e	f								
4	G	H	I	4	g	h	i								
5	J	K	L	5	j	k	l								
6	M	N	O	6	m	n	o								
7	P	Q	R	S	7	p	q	r	s						
8	T	U	V	8	t	u	v								
9	W	X	Y	Z	9	w	x	y	z						
0	0														

Table 53:   Custom Dialpad Characters — Mexican Spanish

BUTTON	NUMBER OF TIMES DIALPAD BUTTON IS PRESSED														
	1	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**	**10**	**11**	**12**	**13**	**14**	**15**
1	:	-	/	,	.	;	(	)	&	+	*	!	?	#	1
2	A	B	C	2	A	b	c	Á	Á						
3	D	E	F	3	D	e	f	É	é						
4	G	H	I	4	G	h	i	Í	Í						
5	J	K	L	5	J	k	l								
6	M	N	O	6	M	n	o	Ñ	Ó	ñ	ó				
7	P	Q	R	S	7	p	q	r	s						
8	T	U	V	8	T	u	v	Ú	Ü	ú	ü				
9	W	X	Y	Z	9	w	x	y	z						
0	¡	¿	ª	º	0										

Table 54:   Custom Dialpad Characters — Canadian French

NUMBER OF TIMES DIALPAD BUTTON IS PRESSED

BUTTON	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**	**10**	**11**	**12**	**13**	**14**	**15**
1	:	-	/	,	.	;	(	)	&	+	*	!	?	#	1
2	A	B	C	2	A	b	c	À	Â	Æ	Ç	à	â	æ	ç
3	D	E	F	3	D	e	f	È	É	Ê	Ë	è	é	ê	ë
4	G	H	I	4	G	h	i	Î	Ï	î	ï				
5	J	K	L	5	J	k	l								
6	M	N	O	6	M	n	o	Ô	Œ	ó	œ				
7	P	Q	R	S	7	p	q	r	s						
8	T	U	V	8	T	u	v	Ù	Û	Ü	ù	û	Ü		
9	W	X	Y	Z	9	w	x	y	z	Ÿ	Ÿ				
0	«	»	0												

Phones Supporting Old Character Bitmaps

All phones other than the Mitel 52xx/53xx IP phones, the 8660 IP phone, and the 8528, 8568, and 8560 digital telephones are unable to support all of the new character bitmaps required to fully support the Canadian French and Mexican Spanish languages. However, they continue to support the limited subset of characters that were available prior to v4.0. When using these other phones, the following charts show which characters the user will enter depending on the number of times each dialpad button is pressed.

Table 55: Custom Dialpad Characters — US English, UK English, and Canadian French

BUTTON	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
1	:	-	/	,	.	;	(	)	&	+	*	!	?	#	1
2	A	B	C	2	a	b	c								
3	D	E	F	3	d	e	f								
4	G	H	I	4	g	h	i								
5	J	K	L	5	j	k	l								
6	M	N	O	6	m	n	o								
7	P	Q	R	S	7	p	q	r	s						
8	T	U	V	8	t	u	v								
9	W	X	Y	Z	9	w	x	y	z						
0	0														

Table 56: Custom Dialpad Characters — Mexican Spanish

BUTTON	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
1	:	-	/	,	.	;	(	)	&	+	*	!	?	#	1
2	A	B	C	2	A	b	c								
3	D	E	F	3	D	e	f								
4	G	H	I	4	G	h	i								
5	J	K	L	5	J	k	l								
6	M	N	O	6	M	n	o	ñ							
7	P	Q	R	S	7	p	q	r	s						
8	T	U	V	8	T	u	v								
9	W	X	Y	Z	9	w	x	y	z						
0	0														

Phones Supporting Japanese Character Bitmaps

The Mitel 52xx/53xx IP phones do not support Japanese characters. However, for all other phones supported by the system, the following chart shows which Japanese characters the user will enter depending on the number of times each dialpad button is pressed.

Table 57: Custom Dialpad Characters — Japanese

NUMBER OF TIMES DIALPAD BUTTON IS PRESSED

BUTTON	1	2	3	4	5	6	7
1	A	I	U	E	O	a	1
2	KA	KI	KU	KE	KO	i	2
3	SA	SHI	SU	SE	SO	u	3
4	TA	CHI	TSU	TE	TO	e	4
5	NA	NI	NU	NE	NO	o	5
6	HA	HI	FU	HE	HO	tsu	6
7	MA	MI	MU	ME	MO	ya	7
8	YA	YU	YO	yu	8		
9	RA	RI	RU	RE	RO	yo	9
0	WA	WO	N	pa	ba	lg	0

DB Programming and Administrative Web Portal Pages

There are certain areas in DB Programming and the Administrative Web Portal pages where system administrators may want to enter information in a language other than English (for example, phone user names). One method is to use a program like Microsoft Word to type out the information in the desired language and then paste it into DB Programming or AWP.

NOTE	DB Programming and AWP do not support Japanese characters. Also, certain phones cannot display all of the new supported character bitmaps (see the previous sections for details).

Caller ID

The character set and language changes needed to support Canadian French may affect incoming or outgoing caller ID information. For example, if the system receives caller ID data using a different character set or using Unicode, it will attempt to convert this data to a character set supported by the system. In some instances, certain characters may not map and display properly on various phones.

TRUNK LANGUAGE

A flag in database programming determines the language that is used by each trunk. This field can be set to any specific language so that the system can support more than two languages. The language choices include:

- Use Primary Language
- Use Secondary Language
- American English
- British English
- Canadian French

- Japanese

- Spanish

If the trunk is programmed for the primary language, callers who reach a Voice Processing application will hear the voice prompts in primary language, unless overridden. If programmed for the Secondary Language, the voice prompts will be in the secondary language, unless overridden. By default all trunks are set for the primary language.

For information about programming the primary and secondary languages for trunks, see "Languages" on **page 406**.

MULTILINGUAL DO-NOT-DISTURB AND REMINDER MESSAGES

The system provides default Do-Not-Disturb and reminder messages in American English, British English, Canadian French, Japanese, and Spanish. However, the messages that use Japanese characters can be reprogrammed only through an Administrator phone. For more information, refer to the *MiVoice Office 250 Phone Administrator Guide,*.

For feature usage instructions, refer to the applicable phone user guide.

The language of the messages seen by users, both when programming their own phones and when calling another phone that is in DND, is determined by the phone's programmed language. That is, if a system's primary language is Japanese, and the phone is programmed for the primary language, the user will see only Japanese messages when programing a message. If a Japanese-programmed phone calls an American English-programmed phone that has selected DND message 02, the user at the Japanese-programmed phone will see the Japanese version of message 02. Among the IP phones only 8660 IP phones can be used for viewing Japanese messages.

When DND or reminder messages are reprogrammed, the system administrator should attempt to keep the meanings for the messages in both lists the same. That is, if the Primary Language DND message 02 is changed to "PAGE ME," a similar message should be programmed for the secondary language DND message 02.

VOICE PROCESSING

When a Voice Processing application receives a call from a phone or trunk, the system tells the application which language is programmed for that device. For example, if the Primary Language is set to American English and the Secondary Language is Japanese, then:

- If a trunk programmed for American English rings-in to an application, the Voice Processing will play American English prompts.

- If a phone programmed for Japanese calls voice mail, the phone user will hear Japanese prompts.

- If a phone programmed for American English receives a call on a trunk that is programmed for Japanese and then transfers the call to voice mail, the caller using the trunk will hear Japanese voice prompts.

- If a Japanese-programmed phone is forwarded to voice mail, a caller on an American English phone or trunk will hear the American English prompts when the call is forwarded. The user-recorded mailbox greeting will be heard in the language in which it was recorded.

When using a Call Routing Announcement application with digit translation, the individual Voice Processing applications assigned to the digits can be programmed to override the device language and provide prompts in one language only. Using this method, a Call Routing Announcement tree can be programmed that offers callers a choice between languages. For example:

- The Call Routing Announcement application could have a greeting that says, "Thank you for calling. For English prompts, press 1. NIHONGO WA, 2 WO OSHITE KUDASAI (for Japanese, press 2)."

- The digit translation for digit 1 would lead to an application that overrides the calling device's programming and uses only American English prompts.

- The digit translation for digit 2 would lead to an application that overrides the calling device's programming and uses only Japanese prompts.

In the preceding example, the digit translations could be nodes that lead to various other language-specific applications. Or, the first level can give more choices, such as American English voice mail or automated attendant and Japanese voice mail and automated attendant. Either way, the individual applications or nodes can be programmed to play only one language, or they can use the device's language, as needed.

MULTILINGUAL FEATURE IN NETWORK OPERATION

For calls across network nodes, if the language does not match between two nodes, the called voice processing system searches for the designated language in its system in the following order:

1. Primary Language

2. Secondary Language

3. American English

4. British English

5. Canadian French

6. Japanese

7. Spanish

8. If the designated language does not exist in the voice processing system, the system uses the default primary language

USING MULTILINGUAL DIRECTORIES

If Multilingual Support is enabled, directory users will be able to enter specific language characters, as needed. Procedures for using the directories are given in "Directory of Intercom, Speed Dial, and Feature Codes" on .

MUSIC-ON-HOLD AND BACKGROUND MUSIC

The Music-On-Hold (MOH) feature intends to make waiting on hold as pleasant as possible, and it assures the holding party that the call is still connected. Users can choose to listen to background music by entering the Background Music feature code (313).

For feature usage instructions, refer to the applicable phone user guide.

To equip the MOH feature, the system must be connected to an external music source. The music source can be a customer-provided radio, a tape or DVD player, or other device plugged into the 1/8-inch MOH port located on the rear of the chassis. To complete installation of the music source, the MOH feature must be enabled in database programming. If music is not desired, the system can be programmed for the following:

- **Silence**: Callers hear no Music-On-Hold.

- **Tick Tone**: Callers hear tick tone.

- **Ringback**: Callers hear ringback.

- **MOH Port**: Callers hear an external music source. This is the default value.

- **File-based MOH**: Callers hear the MOH file selected in DB Programming. For more information, see "File-Based Music-On-Hold (MOH)" on **page 741**.

> **NOTE** A File-based MOH file cannot be used for background music. The Background Music feature always uses the physical MOH connection and not File-based MOH.

NOTICE

There are often broadcast restrictions associated with copyrighted music. Check with the music's original distributor and/or the broadcast source for restrictions and usage concerning background music and music-on-hold. Mitel is not responsible for any illegal or improper use of copyrighted music connected to the system. With European systems, contact PPL (Phonographic Performance Ltd.) for licenses.

In a Mitel system network, each node can support its own music source(s). If a caller on Node 1 is holding for a user on Node 2, the caller hears the music on Node 2. If a caller is holding for a user on Node 1, and the call moves to Node 2 (due to a transfer, forward, or recall), the user stops hearing the music on Node 1 and begins hearing the music on Node 2.

The system can be programmed to determine the music source a caller hears based on the device for which the caller is waiting. By default, the system determines the music source based on the trunk group on which the call resides. Throughout the manual, the term "music" refers to the selected option.

On a MiVoice Office 250 node, the caller hears the music source programmed for **the node trunk group** on the destination node.

Background music is interrupted for calls, pages, phone programming, and ringing.

OFF-HOOK VOICE ANNOUNCE (OHVA)

The OHVA feature allows an intercom caller (either single line or digital telephone) to establish voice contact with the called phone user, even though the user already has a call in progress on the handset. The called party can then talk freely on both connections, using the handset for the original call and the speakerphone for the OHVA intercom call. The caller on the handset call $_{will}$ be able to hear the OHVA call.

For feature usage instructions, refer to the applicable phone user guide.

In a network, intercom callers can establish OHVA calls to phones on other nodes.

A PC Data Port Module (PCDPM) must be installed on a digital display phone to provide the secondary voice path needed for the OHVA feature.

NOTE	IP and SIP phones do not support OHVA.

To enable the OHVA, the following must be completed:

- The OHVA system flag must be enabled. When the system is in the default state, this flag is enabled.

- Each OHVA-equipped phone can be programmed for $_{placing}$ and/or $_{receiving}$ OHVA calls. By default, OHVA is disabled at the station level.

- Other phones and single line phones can be programmed for $_{placing}$ OHVA calls, as desired. Phones without PC Data Port Modules or secondary voice paths, and single line phones cannot receive OHVA calls since they do not have secondary voice path circuitry. By default, OHVA is disabled at the station level.

- PKM 16-equipped digital display phones can be programmed to override the Camp On and OHVA Screening timers and make Immediate OHVA calls. This is a useful option for Attendant stations.

Any combination of OHVA-enabled and -disabled stations is available through DB Programming. For example, all Attendants may be programmed to make, but not receive, OHVA calls, while other stations are allowed to receive, but not place, OHVA calls.

OHVA calls cannot be processed if the secondary voice path or the speakerphone of the called phone is not available. This occurs when the phone has a different OHVA call in progress, has an active data call in progress, is on an active handsfree intercom or outside call, has handsfree disabled, has a call being changed from handset to speakerphone, has a headset enabled, or is in Do-Not-Disturb. Also, OHVA calls are not possible if the caller is placing a private intercom call or has the Ring Intercom Always feature enabled.

NOTE	To place an OHVA call using a single line phone, the Ring Intercom Always feature must be disabled. To disable this feature, lift the handset of the single line phone and dial feature code 377. By default, the Ring Intercom Always feature is enabled.

If you press the MUTE button while speaking to an OVHA caller on the speakerphone, the handset microphone will be muted and the caller on the handset will not hear you or the OHVA caller. When you press MUTE again, the handset microphone is re-enabled.

To place an OHVA call:

1. When placing an intercom call to a busy OHVA-enabled phone, do not hang up. You hear busy signals until the Camp On and OHVA Screening timers expire. If you hear music after the Camp On timer expires, conditions for an OHVA call were not met and you are camped-on.

OFF-HOOK VOICE ANNOUNCE

phone WITH
PCDPM

Ext. 1103

Ext. 1002

ACTIVE
CALL ON
HANDSET

① Caller places a call to ext. 1002 and hears busy signal

② Display shows INTERCOM CALL FROM 1103

③ After 5 seconds, call is established on speaker, if not blocked.
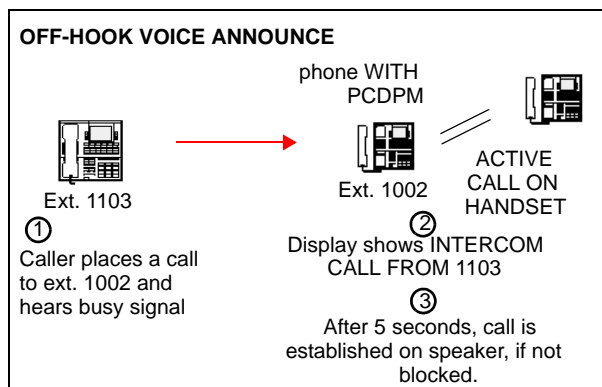
2. When the OHVA Screening timer expires—and if the phone's secondary voice path is available—you are automatically connected to the called party's speakerphone.

To place an immediate OHVA (phones with PKM 16 buttons only):

1. When an intercom call is placed, using a PKM 16 button, to a busy OHVA-enabled phone, do not hang up.

2. Press the PKM 16 button for that phone again. If the phone's secondary voice path is available, you are immediately connected and may speak.

NOTE	This method can only be used if the database has been programmed to allow phones with PKM 16 buttons to bypass the Camp On and OHVA Screening timers.

To receive an OHVA call (OHVA-enabled phones only):

While on a call using the handset, you hear a camp-on tone. The display shows CALL ANNOUNCE FROM *<username>*. Do nothing. When the OHVA Screening timer expires, you hear a double tone. You are connected with the intercom caller via the speakerphone. Your original call remains connected on the handset.

NOTE	If you press MUTE while speaking to the caller on the speakerphone, the handset microphone will be muted and the caller on the handset will not hear you or the OHVA caller. Press MUTE again to enable the handset.

To terminate the OHVA call:

Press the lit Speaker button or have the OHVA caller hang up. If you terminate the original call by hanging up the handset, you remain connected to the OHVA call in the handsfree intercom mode.

To block an off-hook voice announce call (OHVA-enabled phones only):

While on a call using the handset, you hear a camp-on tone. The display shows CALL ANNOUNCE FROM *<username>*. To cause the intercom call to camp on, press the Speaker button. The **IC** button flashes.

To send Do-Not-Disturb signals to the camped-on call:

Press the **DND** button.

ON-HOOK MONITORING

A phone user with a speakerphone can monitor the call (listen to a recorded message, wait for the call to be answered, or wait on hold), and then speak handsfree when answered.

For feature usage instructions, refer to the applicable phone user guide.

OUTSIDE CALLS

When a trunk is selected for receiving or placing an outside call, the voice channel is seized and cannot be used by any other phone—unless the Conference feature is used, as described in "Conference Calls" on **page 282**. If the desired trunk is busy, the phone user can camp on or request to be queued for a callback. They include placing calls on hold, call waiting, call transfer, reverse transfer, conferencing, and call forwarding. For background information, see "Outgoing-Access, Allowed-Answer, and Ring-In Assignments" on **page 186**. For an explanation of trunk groups and the use of the automatic trunk answer and selection feature codes, see "Automatic Call Access" on **page 277**.

PLACING OUTSIDE CALLS

Phones can be programmed to select a trunk, a trunk group, or ARS when a CALL button is pressed. In the default state, all CALL buttons are programmed for ARS access.

On display phones, the dialed number is displayed with hyphens separating the toll field, equal access field, area code, office code, hookflashes, pauses, asterisks, pounds, Centrex codes, and/or absorbed digits. When the system absorbs local trunk digits, the digits are displayed even when they are not dialed. For example, if 423 is absorbed and 6767 is dialed, 423-6767 is displayed.

When placing a call, begin dialing before the Dial Initiation timer expires. If the timer expires, the system drops the trunk connection and sends repeating reorder tones. This timed response prevents a trunk from being tied up accidentally.

For feature usage instructions, refer to the applicable phone user guide.

RECEIVING OUTSIDE CALLS

What you hear when you receive an outside call depends on whether you are using a phone with multiple lines or a single-line phone, as described in the following sections.

OUTSIDE CALL PHONE SIGNALS

You hear one of the following signals when receiving an outside call.

- **Repeating long tones and a CALL button or individual trunk button is flashing at the fast rate**: A call is ringing in. Lift the handset and/or press the flashing individual trunk button, flashing CALL button, or ANSWER button. Phones that have menu buttons with a programmed voice mail extension can press the SEND TO V-MAIL menu button to transfer the call to voice mail. The individual trunk or CALL button flashes slowly during the call. Display phones show <*trunk name*> RINGING IN.

- **Intercom call or page announcing a call and a CALL button or individual trunk button is flashing at the fast rate**: A call has been transferred to your phone. Lift the handset and/or press the flashing individual trunk button, flashing CALL button, or ANSWER button. (Phones with menu buttons that have a programmed voice mail extension have a SEND TO V-MAIL menu button that transfers the call to voice mail.) The individual trunk or CALL button flashes slowly during the call. Display phones show TG XXXX <or *group name*> TFER FROM EX XXXX <*or username*>.

- **An individual trunk button is flashing at the fast rate, there is no ring signal, and you have allowed answer for the trunk. You may hear ringing on another phone**: Press the fast-flashing individual trunk button as described above or enter the Automatic Trunk Answer feature code (350). The ANSWER button cannot be used to answer these calls.

SINGLE LINE PHONE

When receiving an outside call on a single line phone:

- **If you hear a page announcing a transfer and/or repeating double tones**: A call is ringing in. Lift the handset; you are connected to the caller.

- **If you hear a call ringing in on another phone and you have allowed answer**: Lift the handset and enter the Automatic Trunk Answer feature code (350). You are connected to the lowest-numbered trunk ringing in for which you have allowed answer.

PAGING

The Paging feature allows announcements to be made through phone internal speakers. Optional external paging equipment such as amplifiers and paging speakers may be installed using the external PAGE port on the rear of the chassis. For more information, refer to the "Installation" chapter in the *MiVoice Office 250 Installation Manual* .

For feature usage instructions, refer to the applicable phone user guide.

PAGING SETUP

The system supports up to 10 paging zones. When the system is in the default state, all phones are assigned to Paging Zone 1 to provide an All-Page zone. Phones, trunks, and the external paging port(s) can be assigned to any, all, or none of the paging zones, as desired.

NOTE	Placing a large number of phones in a paging zone may affect system performance. If system operation is affected when a page is placed to a particular page zone, remove some phones from that zone or change to external paging for the area served by that page zone.

In a network, all phones and trunks within a page zone must reside on the same node as the page zone. However, a page zone can contain external page ports on other nodes.

NOTE	Because the external paging ports have intercom numbers (91000–91003), phones can place an intercom call to the external paging ports, instead of using the Page feature, to make a page over the external paging speakers.

Pages are not heard on phones that have been removed from paging (using the feature code as described below), are in DND, are ringing, or are in use. Also, background music on phones and on external speakers connected to phone speaker leads is interrupted for pages. If a trunk in a paging zone is unplugged, the page cannot be completed.

If desired, users can program feature buttons to select the paging zones 0–9 or 0–49 using paging access codes 9600–9609 or 9600–9649, respectively.The Page timer limits the length of pages. If it is set to 0, pages are unlimited in length.

ENABLE OR DISABLE PHONE PAGING

A user can prevent the phone from receiving pages or allow it to receive pages using a feature code. If the phone is assigned to more than one page zone, all zones are enabled or disabled. The user cannot toggle between individual zones.

RECORD-A-CALL

IIf your system is programmed with a Record-A-Call application, the phones can be programmed to use the Record-A-Call feature. It allows users to enter a feature code whenever they want to record an ongoing call in their designated Record-A-Call mailbox. Users can retrieve the recorded messages later, just as they would any other mailbox messages.

For complete information about Record-A-Call, see "Record-A-Call" on .

REDIAL

The Redial feature can store one telephone number in redial memory at the phone (manually dialed or Speed Dialed numbers up to 32 digits). If the phone user reaches a busy number, is disconnected, or if there is no answer, the number can be redialed easily. The phone user simply lifts the handset and presses the REDIAL button or enters the Redial feature code. A trunk access code is automatically entered and the telephone number is redialed. (If redialing while still connected to an outside call, the connection is dropped and the trunk is reseized before the number is dialed.)

For feature usage instructions, refer to the applicable phone user guide.

Only one telephone number can be stored in the phone's Redial memory at a time. This number can be stored in one of two ways, depending on phone programming:

- **Last number saved:** The desired number is manually stored in Redial memory by the phone user. Dialing other numbers does not change the stored number. It only changes when a new number is stored.

- **Last number dialed:** The last number manually dialed or Speed Dialed is automatically stored. It changes every time the user dials a telephone number. (This is the default value of the Redial feature.)

The Last Number Saved and Last Number Dialed features work differently depending on the phone status when it is used, as shown in **Table 58**. Individual phone programming determines the mode of the Redial feature (Last Number Dialed or Last Number Saved). The Redial feature code (380) performs the redial function (programmed under the REDIAL button). For more information, see "Phone Feature Codes" on **page 769**. Non-display System Speed Dial numbers cannot be redialed at a display phone.

Table 58:   Phone Responses with Last Number Saved, Last Number Dialed

PHONE STATUS WHEN REDIAL FEATURE IS USED	PHONE HAS LAST NUMBER SAVED	PHONE HAS LAST NUMBER DIALED
Idle phone	Saves the last trunk access code and telephone number that was dialed.	A trunk is selected using the same trunk access code as used to place last outside call, and the last telephone number is dialed.
Intercom dial tone (on or off hook)	Saves the last trunk access code and telephone number that was dialed.	A trunk is selected using the same trunk access code as used to place last outside call, and the last telephone number is dialed.
On an intercom call	Releases the current call, seizes a trunk using the saved trunk access code, and dials the saved telephone number.	Releases the current call, seizes a trunk using the access code used on the last outside call, and dials the last telephone number dialed.
After selecting a trunk but *before* dialing	Redials the saved telephone number on the currently selected trunk rather than the saved trunk access.	Redials the last telephone number dialed, using the currently selected trunk.

Page 1 of 2

Table 58:   Phone Responses with Last Number Saved, Last Number Dialed (continued)

PHONE STATUS WHEN REDIAL FEATURE IS USED	PHONE HAS LAST NUMBER SAVED	PHONE HAS LAST NUMBER DIALED
After selecting a trunk *and dialing* one or more digits	Releases the current call, seizes a trunk using the saved trunk access code, and dials the saved telephone number.	Releases the current call, reseizes the trunk (using the access code used on that call), and redials the digits that were dialed.
On an incoming outside call	Releases the current call, seizes a trunk using the saved trunk access code, and dials the saved telephone number.	Releases the current trunk, seizes a trunk using the same trunk access code used on the last outgoing call, and dials the last number dialed.
Idle phone	Saves the last trunk access code and telephone number that was dialed.	A trunk is selected using the same trunk access code as used to place last outside call, and the last telephone number is dialed.

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REDIRECT CALL

The Redirect Call feature (feature code 331) allows the user to route ringing outside, intercom, and camped on calls to another phone, hunt group, or outside number. This capability is in addition to the option of redirecting calls to voice mail or DND. Routing redirected calls is subject to toll and trunk restrictions and does not require any software license features.

For feature usage instructions, refer to the applicable phone user guide.

The call types listed below do not follow Call Forwarding and cannot be redirected:

- Agent Help Request calls

- Queue Callback calls

- Recalls

In addition, calls may not be redirected to the following destinations:

- Restricted outside numbers

- Phones in Do-Not-Disturb mode

- Phones that are forwarded

- Another phone's mailbox

- The original phone (display phones show CALL FORWARDED TO SOURCE)

- An unreachable destination

If an ARS, trunk group, or trunk number is entered, the system prompts the user to enter the destination telephone number. Calls may not be redirected to a node number, but may be redirected to a phone on another node by entering the correct extension number.

If the applicable Forwarding timer expires before the user completes the redirection process, the system terminates the call.

REMINDER MESSAGES

Reminder messages are set to signal a phone at a specified time. The user can select one of 20 different messages and set the reminder time up to 24 hours in advance. These messages can be reprogrammed by the system administrator or by using an administrator phone.

For complete information about Reminder Messages, see "Reminder Messages" on **page 466**.

REMOTE PROGRAMMING

The Remote Programming feature allows a user to place a phone in DND mode or forward the phone's calls, either from another phone or through DISA. Each phone has a passcode to limit access to this feature. For information about DISA, see "Using DISA" on .

For feature usage instructions, refer to the applicable phone user guide.

REMOTE PROGRAMMING IN A NETWORK

In a network, Remote Programming can be performed on one node for any phone that is programmed as an off-node device on that node. You must go through DISA for remote programming of any phones that are not programmed as off-node devices. For off-node device programming instructions, see "Creating Off-Node Devices" on

REMOTE PROGRAMMING PASSCODE

The phone passcode is used only for the Remote Programming feature.

NOTE	To prevent unauthorized use of the Call Forward feature, all phones using Remote Programming should have a passcode. Difficult-to-guess passcodes should not match the extension number or consist of a single digit repeated several times.

The phone passcode can be up to 8 digits in length. The default passcode is the extension number of the phone. The passcode can be changed by entering the Program Passcode feature code at the phone or when using the Remote Programming feature. It can also be programmed through Individual Phone programming.

If the passcode is changed from a phone, the user is prompted for the old passcode, then the new passcode, and then asked to verify the new passcode. If it is programmed through Remote Programming, the user has already entered a correct passcode. So, the user is only prompted for the new passcode and prompted to verify the passcode.

If the verified passcode and new passcode not match, the old passcode is retained and the programming session is canceled.

SPEED DIAL

The system provides the following Speed Dial features:

- **System Speed Dial**: Speed Dial allows users to dial stored telephone numbers quickly. Up to 1000, 48-digit System Speed Dial numbers can be stored in system memory. If desired, an identifying name can also be stored with each number. Phones programmed with access to this feature can dial any of the numbers on the list. The Administrator maintains the System Speed Dial list. For complete information about System Speed Dial, see "System Speed Dial" on **page 470**.

NOTE	Only 500 System Speed Dial numbers can be programmed manually. 1000 System Speed Dial numbers can be imported at a time.

- **Station Speed Dial**: A phone user can program up to 10 Station Speed Dial numbers of 16-digits each. Single line phones use Speed Dial location codes (0–9). Display phones use Speed Dial buttons, if programmed in the phone's keymap, or location codes 0–9. For complete information about Station Speed Dial, "Station Speed Dial" on **page 477**.

- **System Directory (Intercom and Outside)**: The intercom directory enables display phone users to look up intercom extension numbers and user names. The outside directory enables display phone users to look up System Speed Dial numbers and associated names. This capability requires the Directory feature. See **page 290**.

For feature usage instructions, refer to the applicable phone user guide.

TRANSFER – CALL TRANSFER

The following sections provide information on the Transfer feature. For feature usage instructions, refer to the applicable phone user guide.

FEATURE CODES

There are two feature codes for transferring intercom and outside calls to other phones, Voice Processing applications, hunt groups, off-node devices, or outside telephone numbers. The call transfer options are as follows:

- **Transfer to ring**: You can transfer intercom or outside calls to another phone, a Voice Processing application, a hunt group, or an outside telephone number.

- **Transfer to hold**: Either intercom or outside calls can be transferred to another phone and placed on hold using this feature.

TRANSFERRING CONFERENCE CALLS

| **NOTE** | For information on screened and announced transfers from automated attendants, see **page 279**. |

An established conference can be transferred to a phone. While the transfer is taking place, the parties in the conference remain connected to each other and may converse. The transfer appears at the destination phone in the same manner as any other transferred call along with a CONFERENCE TFR FROM *<username>* display and may be answered by the phone user.

TRANSFER TO A SYSTEM FORWARD

When a phone user transfers a call to the principal phone of a system forwarding path, the Handsfree Announce System Forward system-wide flag determines what the caller hears. If enabled, the caller will place a handsfree call to the principal phone and can announce the transfer. The call does not enter the forwarding path. If the option is disabled, the transferring phone user hears ringing and can announce the call only if it is answered by the principal phone or one of the forwarding points. For more information about system flags, see "System Flags" on **page 751.**

TRANSFERS AND OUTSIDE CALLS

If a phone user answers another call while a party is waiting on Transfer Hold, the waiting party is moved to Individual Hold at that phone. If the Hold timer expires while a call is on Transfer Hold, the call recalls the phone. The phone user must return to the call on Hold and begin the transfer process again.

Depending on the quality of the trunks being used, some reduction in voice volume may occur when an outside call is transferred to an outside telephone number.

TRANSFER TIMERS

The following five timers affect the Call Transfer feature:

- Transfer Attendant
- Transfer Available
- Transfer Busy
- Transfer Voice Processor
- Unsupervised CO

See "Timers and Limits" on **page 756** for details about these timers.

TRANSFER-TO-CONNECT ALLOWED PHONE FLAG

If the Transfer-To-Connect Allowed flag is enabled, an announced transfer call (including an announced transfer from voice mail) connects immediately with the destination once the transferring party hangs up. (If the flag is disabled, the recipient must press a CALL button to answer the transferred call). See "Transfer-To-Connect Allowed" on **page 422** for details.

TRANSFER TO RING

Transfer to Ring transfers calls to other phones.

TRANSFER TO HOLD

A call transferred to hold at a phone does not ring or send a display message while holding. After the Hold timer expires, the phone rings or sends call waiting signals. Also, calls transferred to hold do not recall the transferring party; they recall the receiving party's attendant if unanswered after the Hold and Recall timers expire.

TRANSFER RECALLS

If a call transferred to another phone—not to hold—is not answered before the appropriate Transfer timer expires, the call recalls the transferring phone's recall destination, defaulting to the transferring phone and the Recall timer starts. The call rings until the Recall timer expires. If unanswered, it recalls the transferring party's attendant and the Abandoned Call timer starts. If the transferring phone has no attendant, the call continues to recall at the transferring phone. If the call is not answered before the Abandoned Call timer expires, the call is disconnected by the system.

In a network, a phone's transfer recall destination can be an off-node device.

TRANSFER – REVERSE TRANSFER

Calls on hold or calls ringing or recalling at a phone or a hunt group can be picked up by any other phone with the Reverse Transfer feature. A user can reverse transfer a call from a phone location or a hunt group on another node.

If more than one call is ringing or holding at the phone or hunt group, a priority list determines which call is reverse transferred. Calls are selected in the following order, and if more than one call of the same type is at the phone, the calls are picked up in the order they were received:

1. Ringing calls

2. Camped-on calls

3. Holding outside calls

4. Holding intercom calls

If a call is reverse transferred from a hunt group announcement or overflow station, and a phone in the hunt group becomes available, the call will be disconnected from your phone immediately when answered by the hunt group.

Group Call Pick-Up: A call can be reverse-transferred from a phone within a hunt group, using the hunt group's extension number, even if the call was not a hunt group call. To reverse transfer the call, use the phone's extension number or the extension number of its hunt group. See "Group Call Pick-Up" on **page 568** for details.

There is a programmable phone flag called "Transient Call Indication On Call Answer." This flag determines whether the phone user sees a call indication display when answering a call by pressing a secondary extension button or by reverse transferring. If enabled, the display indicates if the call was ringing, recalling, transferred, or holding at the other phone. In the default state, it is enabled.

USER-PROGRAMMABLE FEATURE BUTTONS

Individual phone buttons can be designated as "user-programmable" in the keymaps during database programming. The system administrator can program the default feature codes, and the phone user can reprogram the feature codes as desired. For programming instructions, see **page 427**.

Secondary extension buttons can be assigned to user-programmable buttons during keymap programming. If so, the user can determine the primary phone that is associated with the secondary extension button. For programming instructions, see "Programming Phone Keymap Buttons" on **page 439**.

For feature usage instructions, refer to the applicable phone user guide.

ASSISTANTS

The following assistants are available for end users:

- **Configuration Assistant**: The Configuration Assistant is a voice guided configuration portal that provides easy-to-use, remote access to the following end-user phone configuration options (see "Configuration Assistant" on **page 458** for details):
  - Dynamic Extension Express
  - Do-Not-Disturb (DND)
  - Call Forwarding
  - Meet-Me Conferencing
  - Administrator Functions

- **Conference Assistant**: The Conference Assistant is a device with a dialable extension that users will call to access Meet-Me Conferences. See "Conference Assistant" on **page 462** for details.

- **OfficeLink Assistant**: See OfficeLink Assistant: "OfficeLink Assistant" on **page 462** for details.

MITEL 53XX IP PHONE DESKTOP APPLICATIONS

This section contains the following information:

- **Feature Description** below

- Programming Desktop Applications for 53xx IP Phones on **page 344**

FEATURE DESCRIPTION

With the v5.0 or later software release, Mitel 53xx IP phones can be equipped with a variety of additional desktop applications. **Table 59** shows supported desktop applications and IP phones.

Table 59:   New Desktop Applications for Mitel 53xx IP Phones

DESKTOP APPLICATION	SUPPORTED IP PHONES:			
	5320	5330	5340	5360
Call History	✓	✓	✓	✓
People			✓	✓
HTML	✓	✓	✓	✓
Language	✓	✓	✓	✓

For specific end-user instructions on using these desktop applications, refer to the appropriate phone user guides.

CALL HISTORY APPLICATION

The Call History application allows 5320, 5330, 5340, and 5360 IP phone users to display a list of the calls that were missed, made, and answered. The list shows the most recent 50 calls. After the 50-call limit has been reached, the oldest call record is replaced each time a new call is missed, made, or answered.

From this application, user's can:

- display calls by type

- display call information (if available) for each call

- return missed calls by pressing a button

- delete or modify digits before saving or dialing a call

- delete calls

The Call History application leverages the system's call logging feature. If call logging is disabled in DB Programming, the Call History app will no longer be updated. If the phone user clears the call logs list and the phone resets, the Call History application will be empty.

Also, because there are two ways to view missed calls on 5320/30/40/60 phones (the new Call History application and the existing call logging feature), there are cases where accessing the missed calls lists will not synchronize the two views. The Display Missed Calls On Phone phone flag mitigates issues with the "out-of-sync" missed calls information (see page **page 420**). Mitel recommends setting this flag to No for any 53xx users that use the Call History application.

See **page 344** for programming instructions.

PEOPLE (CONTACTS) APPLICATION

The People application allows Mitel 5340 and 5360 IP phone users to add, delete, and edit contact names and numbers. Users can also dial anyone on the contact list from the People application. The People window displays up to eight contacts per page, ordered alphabetically by last name.

To allow the People application, the phone's extension number must be assigned as the main extension of a "User" in the Users folder within DB Programming. See **page 344** for additional programming instructions.

HTML APPLICATIONS

Mitel 5320, 5330, 5340, and 5360 IP phones may be equipped with various HyperText Markup Language (HTML) applications available for customized use. (Currently this is limited to customized screen saver-type applications only.)

The Mitel HTML Desktop Toolkit, which is available on Mitel Online, enables simple, intuitive development of customized applications, such as customer-specific "branded" screen savers, that are easily integrated with telephony functions of certain Mitel IP phones. Customers or developer partners requiring access to Developer Support on the HTML Desktop Toolkit must join the Mitel Solutions Alliance (MSA). All MSA member levels include support on the HTML Toolkit. More information and the MSA online application form are available at www.mitel.com/msa.

With v5.0 or later, DB Programming uses an associated Applications profile to determine which HTML applications will be available on which Mitel IP phones.

Besides the resident 5360 screen saver and help applications available prior to v5.0, the system now includes a 5320/30/40 screen saver application.

See **page 344** for programming instructions.

LANGUAGE APPLICATION

The Language application allows 5320, 5330, 5340, and 5360 IP phone users to select the language in which phone prompts and applications appear. Supported languages are American English, British English, Canadian French, and Mexican Spanish.

The system can support all four languages at one time. The language choices appearing on your display are controlled by system programming.

There is no programming required to support the Language application.

NOTE	Mitel recommends that 5320/30/40/60 IP phone users use the Language application to switch languages, rather than using the Change Language feature code (default is 301) to switch between the Primary and Secondary language. The Language application provides a nicer visual interface and gives the user more potential language options.

PROGRAMMING DESKTOP APPLICATIONS FOR 53XX IP PHONES

The following section provides instructions for programming desktop applications for 53xx IP phones.

PROGRAMMING CALL HISTORY AND PEOPLE APPLICATIONS

If desired, phone feature buttons can be assigned to provide direct access to the People and Call History applications so that users do not have to first go through the Applications menu. (Note that with this release, the Language application cannot yet be assigned to a feature button.)

To support the Call History and People desktop applications on a 53xx IP phone, you can assign a keymap ID that has the following key types as programmable buttons on a 53xx IP phone:

- **Application -- All**: *(For the 5320/30/40/60 keymaps.)* This key type gives access to the top-level Applications menu that contains all of the desktop applications.

- **Application -- Call History**: *(For the 5320/30/40/60 keymaps.)* This key type gives access to the Call History application menu.

- **Application -- People**: *(For the 5340/60 keymaps.)* This key type gives access to the People application menu.

See "Keymap Value Column" on **page 427** for details about key types.

To program a desktop application on a 53xx IP phone keymap:

1. Select System – Phone-Related Information – Key Assignments – **IP/Digital Telephone**.

2. Right-click and select **Add to IP/Digital Telephone List**. Follow the instructions in the DB Programming Help to add a keymap ID and configure the programmable button(s) for the Application -- Call History, Application -- People, and/or Application -- All key types.

3. In the System\Devices and Feature Codes\Phones*<53xx phone>*\Keymaps folder, assign the keymap ID you just configured for the 53xx IP phone that you want to apply. Refer to the DB Programming Help for programming details.

> **NOTE** To support the People desktop application on a 53xx IP phone, the phone user's extension must be assigned to the main extension of a User.

To assign the phone extension to the main extension of a User:

1. Select Users – *<User>*.

2. Set the Main Extension to use the extension of the phone.

PROGRAMMING HTML APPLICATIONS

This section describes how to program and manage HTML applications. See "HTML Applications" on **page 343** for details about HTML applications.

To determine which HTML applications will be supported by which 53xx IP phones:

For an Online Session

1. Select System – Phone-Related Information – **Applications**.

2. Right-click and select **Manage Application Files**. A View/Program the HTML Application Files Currently Loaded on the System window, similar to the one below, appears. The window displays information on all of the applications uploaded to the system, including application file names, application names, type, size, and supported phone models, language, and application programmed. The application programmed column lists the ID of the application that is programmed in the database matching the name of the application in the application file. In following example, if application ID 1 was programmed in DB Programming with name "Default_ScreenSaver," the first and third files would show "1" for this column.

NOTE	Application files must be uploaded/deleted through the MiVoice Office 250 Administrative Web Portal (AWP) prior to connecting with DB Programming to ensure an accurate list of application files.

3. Select files with no application programmed, and then click **Add**. The Application Programmed column is updated according to the IDs that are programmed in the System\Phone-Related Information\Applications folder.
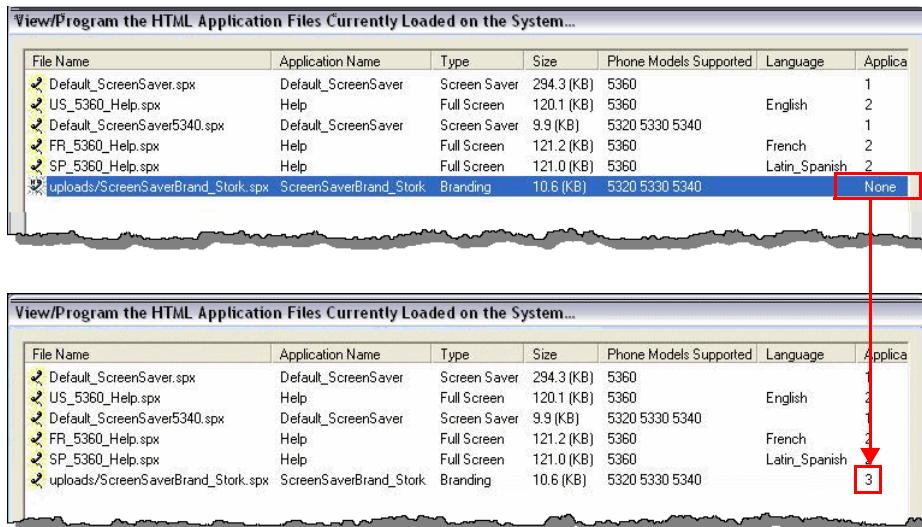
4. Add the newly programmed applications to Application Profiles as described in "Assigning HTML Applications to the Application Profile" on **page 347** so that phones can use them in the System\Phone-Related Information\Application Profiles\X\Applications folder.

For an Offline Session

1. Select System – Phone-Related Information – **Applications**.

2. Right-click and select **Create Application**. A new application is added to the Applications list.

3. Manually program the following fields for the application:

    •   **Name**: Type the name of the application as specified in the manifest file. The name can contains up to 25 printable characters, except for ', |, and '.

    > **NOTE** Names are expected to be unique. If different files have the same application name, they will be assumed to be used for the same HTML application. Duplicate names for different applications are *not supported*.

    •   **Type**: Select the application type. The supported application types are Branding (customer-specific branded screen saver), Full Screen (help), and Screen Saver (default screen saver). It is set to Undefined by default. This field is only used for enforcing type restrictions when adding to a profile. The MiVoice Office 250 relies on the type specified in the application file, not the type programmed, for determining the application type.

    •   **Label**: (This field is reserved for future use.) Type the label for the application (up to 15 characters) as it appears on the phone's display.
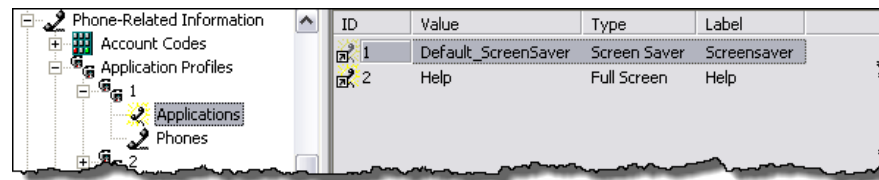
To assign HTML applications to 53xx IP phones:

Creating Application Profiles

1.   Select System – Phone-Related Information – **Application Profiles**.
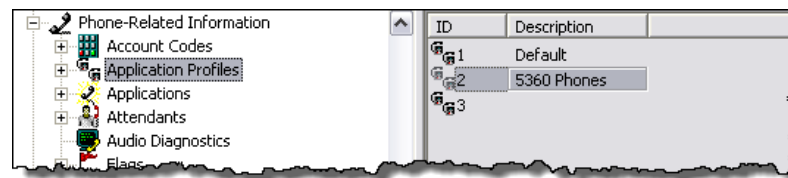
    The Application Profiles folder determines which HTML applications will be available on which Mitel IP phones. There is one default profile containing the two default applications. All 52xx/53xx IP phones are associated with this profile by default so they may receive the

Screen Saver and Help default applications. The only additional HTML application currently supported is a Branding Screen Saver.

2. *(Optional)* To add an Application Profile (up to 25), right-click and select **Create Application Profile**. The Get ID dialog box appears. Follow the instructions in the DB Programming Help to create an ID. When finished, a new Application Profile is added to the Application Profiles list.

3. Type the description of the Application Profile (for example, 5360 Phones).

4. *(Optional)* To remove the Application Profiles from the Application Profiles list, select the Application Profile(s), and then right-click **Delete Selected Application Profile(s)**.

Assigning HTML Applications to the Application Profile

1. Select System – Phone-Related Information – **Application Profiles.**

2. Click the Application Profile you just created, and then click **Applications**.

3. Right-click and select **Add To Applications List**. When the Add To Applications List dialog box appears, click **Next**. The HTML applications that are programmed in the System\Phone-Related Information\Applications folder appear in the box.

4. Select the HTML application types that you want to add, and then click **Add Items**. When the session is online, you can program one default Screen Saver application and up to three Branding Screen Saver applications for the entire Application Profiles in the system. Also, if the select applications have been uploaded to the system, but are not yet programmed in DB Programming, new applications will be created as well as added to the Application Profile.

5. Click **Finish**. The selected HTML applications are added to the Applications list.

6. *(Optional)* To remove the HTML applications from the Applications list, select the application(s), and then right-click **Remove selected Items**.

Assigning Phones to the Application Profile

1. Select System – Phone-Related Information – **Application Profiles.**

2. Click the Application Profile you just created, and then click **Phones**.

3. Right-click and select **Move To Phones List**. When the Move To Phones List dialog box appears, click **Next**.
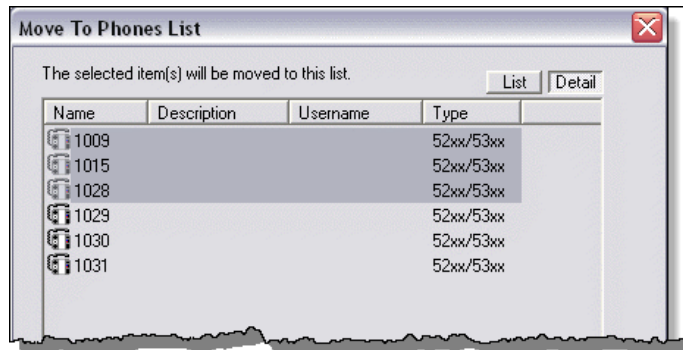
4. Select the phones that you want to add, and then click **Move Items**. Note that DB programming does not distinguish between the various 52/53xx phones; therefore, any programming related to applications allows any 52/53xx phones to be included. This includes placing any of these phones in the default Application Profile.

5. Click **Finish**. The phones are added to the Phones list.

TROUBLESHOOTING END USER FEATURES

Table 60 summarizes the troubleshooting strategies recommended for resolving system features discrepancies.

Table 60:   System Features Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Feature does not work correctly	User error	Refer to the applicable phone user guide for feature instructions. Also, ensure that the feature is available on the software package installed.
	Programming error	Check feature code programming. Also, check user-programmable feature key programming.
	Defective phone	Replace the phone and/or perform the phone self-test as described in the Installation chapter in the *MiVoice Office 250 Installation Manual* .
	Defective Digital Endpoint Module	Replace the associated module.
	Defective processor module	Replace the module if faulty.
Cannot transfer CO or intercom calls to other phones	User error (for example, wrong feature code)	Refer to the applicable phone user guide for feature instructions
	Called phone is in Do-Not-Disturb	A phone in Do-Not-Disturb cannot receive transferred calls.
Cannot transfer calls to outside numbers	User error (for example, wrong feature code)	Refer to the applicable phone user guide for feature instructions
	Programming error	Check trunk access and toll restriction.
	Defective processor module	Replace the module if faulty.
Cannot transfer incoming CO calls or place them on hold	User error	Refer to the applicable phone user guide for feature instructions.
	User error (trunk button being pressed after initial connection is made)	Set the CO Reseize timer to a higher value. The default value is 3 seconds. Or, program the phone to disallow CO Reseize. See "Timers and Limits" on **page 756** and "Phone/Phantom/Hot Desk Profile Flags" on **page 417**.
	Programming error	Check user-programmable feature button programming.

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Table 60:   System Features Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Cannot initiate a conference	User error	Refer to the applicable phone user guide for feature instructions.
	System capacity exceeded	See the maximum system capacities in the Specifications chapter in the *MiVoice Office 250 Installation Manual* .
	Defective phone	Replace the phone and/or perform the phone self-test as described in the Installation chapter in the *MiVoice Office 250 Installation Manual* .
	Defective Digital Endpoint Module	Replace the associated Digital Endpoint Module.
	Defective processor module	Replace the module if faulty.
Last number redial feature inoperative	User error	Phone may be programmed for last number saved.
	System speed-dial number identified as non-display	A System Speed Dial number identified as non-display cannot be redialed.
	Defective phone	Replace the phone and/or perform the phone self-test as described in the Installation chapter in the *MiVoice Office 250 Installation Manual* .
	Defective processor module	Replace the module if faulty.
Cannot initiate a page	User error	Refer to the applicable phone user guide for feature instructions.
	All phones in the paging zone are busy, or a voice channel is not available	Reorder tone is heard. Wait several seconds and then attempt to place the page again. Paging requires a voice channel.
	All phones in the paging zone are in Do-Not-Disturb	Reorder tone is heard if all phones in the zone are in Do-Not-Disturb and if external paging for the zone is disabled.
	No phones are programmed to receive pages	Reorder tone is heard. Check paging assignment. See "Page Zones" on **page 747**.
	Defective phone	Replace the phone and/or perform the phone self-test as described in the Installation chapter in the *MiVoice Office 250 Installation Manual* .
	Defective processor module	Replace the module if faulty.
House Phone is not working properly or is inoperative	User error	Incoming calls take precedence over outgoing calls.
	Programming error (data base)	Ensure that the phone is designated as a House Phone. Also, make sure the phone has been assigned the correct COS and trunk access. See "Programming Phone Options" on **page 391**.
	Programming error (data base or speed-dial)	Ensure that the correct numbers are in appropriate day number and night number (Speed Dial) locations.

Table 60:   System Features Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Cannot initiate a call forward	User error	Refer to the applicable phone user guide for feature instructions.
	User attempting illegal forward	Phones are not allowed to set call forward if it forms an unconditional loop, the receiving phone is in Do-Not-Disturb, or an invalid intercom number is dialed. COS and outgoing access are checked when a call is forwarded to an outside telephone number. Also, ARS cannot be used to forward to an outside number.
	Defective phone	Replace the phone and/or perform the phone self-test as described in the Installation chapter in the *MiVoice Office 250 Installation Manual* .
	Defective processor module	Replace the module if faulty.
Calls will not forward	User error	Refer to the applicable phone user guide for feature instructions.
	Illegal forward	Conditional forwards (for example, if busy, if unanswered) may form an undetected loop. If a call forward request forms a conditional loop, the call returns to the first phone.
	Defective Digital Endpoint Module	Replace the associated module.
	Defective processor module	Replace the module if faulty.
Phone is not receiving hunt group calls	User error	Hunt group calls may have been halted using the Hunt Group Remove feature code. Or, the phone may be in Do-Not-Disturb.
	Programming error	Check hunt group programming. See "Hunt Groups" on **page 552**.
Phone is not receiving pages	User error	Pages may have been halted using the Page Remove feature code. Or, the phone may be in Do-Not-Disturb.
	Programming error	Check page zone programming for the phone. See "Page Zones" on **page 747**.
Phone cannot be placed in Do-Not-Disturb mode	User error	Refer to the applicable phone user guide for feature instructions.
	Programming error	Phone is programmed to disallow Do-Not-Disturb. See the "Programming Phone Options" on **page 391**.
	Defective phone	Perform the phone self-test as described in the Installation chapter in the *MiVoice Office 250 Installation Manual* .

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Table 60:   System Features Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
The supervisor cannot silent monitor a call	The supervisor is not in the member's hunt group.	Program the appropriate phone as the hunt group supervisor in DB programming.
	The agent is on a peer-to-peer (P2P) call.	Program the agent in a network group that does not allow P2P calls.
	The agent is on a conference call.	The agent must terminate the conference call.
	The agent is receiving agent help.	Wait until the agent help hangs up the call.
	Another supervisor is currently joining the hunt group member's call.	Wait until the other supervisor hangs up the call.
A supervisor's silent monitor terminates	The hunt group member has requested agent help.	None. The supervisor has terminated the call. Due to the agent requesting help the silent monitor has terminated.
	The hunt group member puts the call on hold.	None. The supervisor has terminated the call. Due to the extension\agent putting the call on hold the silent monitor is terminated.
	Another supervisor was silent monitoring and joined the call.	None. The supervisor has terminated the call. If another supervisor has joined the supervised call e.g. a 'Barge In', all other supervisors who are silent monitoring the call will be dropped. If a supervisor has joined the call, any other silent monitor attempts will be refused.
	The call was transferred.	None. A silent monitor on a transfer announcement call terminates after the system completes the transfer.
The user cannot access feature code 320.	The Audio Diagnostics phone flag is not enabled.	Enable the Audio Diagnostics phone flag (under System\Devices and Feature Codes\Phones\<*Node*>\<*phone*>\Flags).
The user does not see the Record-A-Call menu after activating the Audio Diagnostics feature.	The Record-A-Call application does not exist.	Create a Record-A-Call application if one does not exist.
	The Record-A-Call application is not active.	Make sure the Record-A-Call feature is enabled in DB programming.
	A supervisor is silent monitoring a call.	None. The Record-A-Call feature cannot be used on certain calls (such as Agent Help, Station Monitor, Conferences, Paging, and so forth).

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Chapter 8

PHONES AND DEVICES

INTRODUCTION

This section provides information about how to program phones and devices. Phones and devices can either use analog, digital, or Internet Protocol (IP) transmission lines.

Because IP devices require configured network settings, you must program IP phone settings in DB Programming. See "Creating Local IP Phones and Devices" on **page 355**. IP devices also require specific IP configurations and system settings. For more information about IP system settings, see "System and Device IP Settings" on **page 599**.

For phone-related troubleshooting issues, see **page 483**.

VIEWING SYSTEM PHONES

In the Phones programming section, you can create off-node devices for phones on the other nodes and program individual phones on the Local node.

To view the list of phones on the Local node:

1. Select System – Devices and Feature Codes – **Phones**.

2. Double-click **Local**. A list of current phones appears. At this level, you can:
   - Change the extension number
   - Change the description or username
   - View the Phone type (this cannot be changed)
   - Create an 86xx, IP Single Line Adapter, 8602 Softphone, 52xx\53xx, or MiCollab Client MiNET Softphone
   - Create SIP phones:
     - Mitel Unified Communicator (UC) Express SIP Softphone
     - MiCollab Client SIP Softphone
     - Mitel 5610 Cordless Handset and IP DECT Stand
     - MiVoice Conference and Video Phone
     - IP DECT 5603/5604/5607
     - Mitel Phone Manager Softphone
     - 3rd Party SIP Phone (please refer to the Mitel SIP Center of Excellence)
   - Create IP phones from file
   - Convert Digital Phones to 52xx\53xx IP Phones.

CREATING (ADDING) DEVICES

You can create new phones for the local node or remote nodes. You can also use the Configuration Wizard to create phones. See "Launching the Configuration Wizard" on **page 59** for instructions on how to launch the wizard. For complete information about the Configuration Wizard, refer to the *MiVoice Office 250 DB Programming Help*.

Each node in the network has its own extension and feature code programming. However, the network should have a universal numbering plan so that extension numbers on the various nodes do not overlap and do not conflict with feature codes. That is, when planning the extension numbers for each of the nodes in the network, set aside a block of extension numbers (for phones, hunt groups, voice processor applications, and so forth) for each node.

This section contains the following information:

- **Adding Digital Telephones** below
- Creating Local IP Phones and Devices on **page 355**
- Creating Phones from CSV Files on **page 358**
- Creating Off-Node Devices on **page 367**
- Using the Wildcard Character in Off-Node Extensions on **page 368**

ADDING DIGITAL TELEPHONES

For instructions to add digital telephones, refer to the *MiVoice Office 250 Installation Manual* .

CREATING LOCAL IP PHONES AND DEVICES

You can create IP phones, IP softphones, or IP single line adaptors (SLAs) for the local node. When you create any IP phone or device, you must also configure IP phone settings. For more information about IP device settings, see "Phone and Device IP Settings" on **page 641**.

WARNING

Possible Delay in Local Emergency Response to Remote Sites.

IP and SIP phone users should be alerted to the following hazardous situations:

• If an Emergency Call phone number is dialed from an IP or SIP phone located at a remote site that is *not* equipped with a correctly configured gateway, the call will be placed from the location where system chassis is installed rather than from the location where the emergency call is made.

In this situation, emergency responders may be dispatched to the wrong location. To minimize the risk of remote site users misdirecting emergency responders, Mitel recommends regular testing of MGCP/SIP gateway trunk(s) for dial tone.

• If uninterruptible power supply (UPS) protection has *not* been installed as part of the MiVoice Office 250, IP and SIP phones will *not* operate when electrical power fails either at remote sites or at the main system location.

To place calls during a power failure in this situation, IP and SIP phone users can only use a single line phone connected to one of the power failure bypass circuits built into the system chassis. If a phone connected to a power failure bypass circuit is not available, users should make emergency calls **from a local phone *not* connected to the system**. For more information about the Power Bypass feature, refer to the *MiVoice Office 250 Installation Manual .*

NOTES	You can oversubscribe the IP resources by configuring more devices than can be active at the same time. Although you can create more IP phones or IP trunks than resources reserved, you *cannot* create more IP devices than the system supports. If you attempt to create more IP phones or IP trunks than the available value, an error message appears stating that you cannot exceed the system limit.

Fax over IP (FoIP) and modems are not supported through an IP SLA. To connect a fax machine, attach it to a single line port on one of the single line interfaces supported by the MiVoice Office 250. Mitel currently supports T.38 FoIP only. |

To create (add) devices:

1. Select System – Devices and Feature Codes – **Phones**.

2. Right-click anywhere in the right pane to show device options.

3. Select one of the following device options:

   • **Create 52xx/53xx**: For all 52xx/53xx-series phones.

   • **Create 86xx**: For all 86xx-series hard IP phones.

   • **Create IP Single Line Adapter**: For IP single line adaptors.

   • **Create 8602 Softphone**: For 8602 Softphones.

   • **Create SIP Phone**: For SIP phones. See for details.

   • **Create MiCollab Client MiNET Softphone**: For a MiCollab Client MiNET Softphone. See for details.

   • **Create IP Phones from File**: Create phones from a comma-separated value (CSV) file. See "Creating Phones from CSV Files" on for details.

   One of the following dialog boxes appears, depending on your selection.
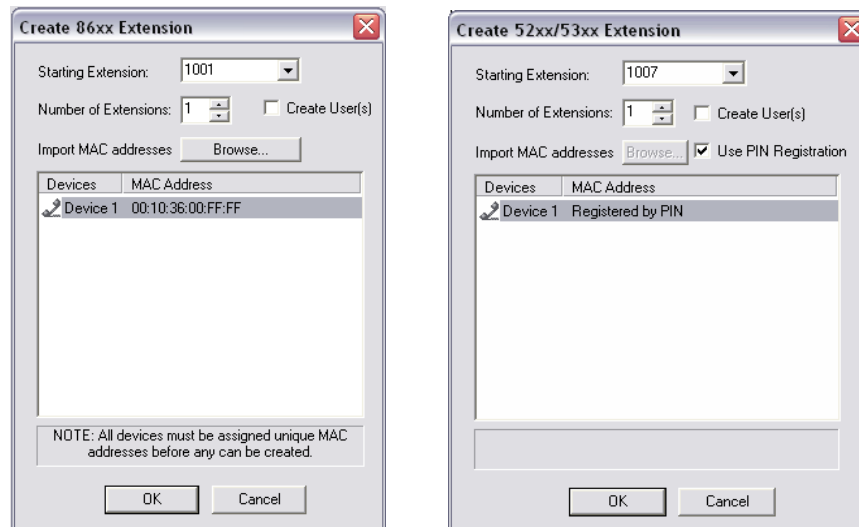
4. From the **Starting Extension** list, select a starting extension from the list of available extensions, or type the extension number.

5. From the **Number of Extensions** list, select or type the number of extensions that you are creating.

6. Select **Create User(s)** if desired. See "Auto-Creating Users when Phones/Phantoms are Created Manually" on **page 792** for details.

7. *(For 52xx/53xx phones only)* Select **Use PIN Registration** if desired. See **page 363** for details.

8. Do one of the following:

   • *If you are creating IP phones or IP SLAs*:

   Click any phone with a red "X" and type a valid MAC address. You can also click **Browse**, and then import the MAC addresses from a batch file. The batch file can be a simple text file consisting of a list of MAC addresses. If you need only 20 addresses from the list, the first 20 addresses are imported.

   • *If you are creating 8602 softphones*:

   Click the device value to enter an ID for the 8602 Softphone. For troubleshooting purposes, use the extension number as the last digits of the device ID. For example, if the extension number is 1001, the device ID could be 86.02.36.00.**10.01**.

9. After you have entered all of the MAC addresses, click **OK**. If you entered the same number as an existing extension, an error message appears and you must enter a new number. The new off-node device appears in the list without a description or username.

> **NOTE** Make sure that you program a speaker button for users who want to use a headset on the 5304.

To create a MiCollab Client MiNET Softphone:

1. Select System – Devices and Feature Codes – **Phones**.

2. Right-click anywhere in the right pane.

3. Select **Create MiCollab Client. . .**.

4. When the Create MiCollab Client Extension dialog box appears, select the starting extension and the number of phones of this type to create.

5. Click **OK**.

6. If you attempt to create a MiCollab Client MiNET Softphone with an extension that causes a conflict with another device extension, the following dialog box appears. Select the conflicted phone in the Name column, and then type a different extension or select an extension from the drop-down list.
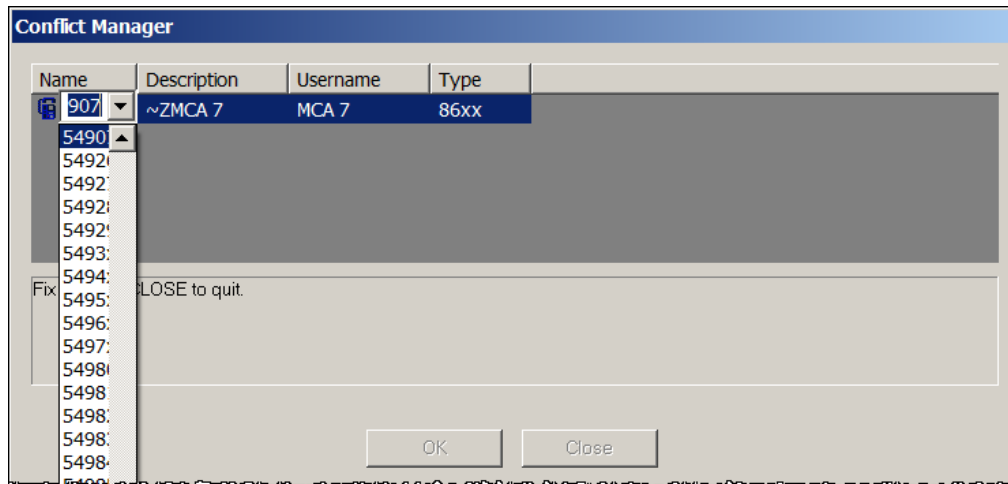
7. Click **OK**, or click **Close** without making changes.

NOTE	When you program the UC Advanced Softphone, you do not need to manually type a MAC address. Instead this address is automatically populated by DB Programming. Just as with other IP phones the MAC address is shown under the Network Configuration folder for that phone (under System\Devices and Feature Codes\Phones\<*phone*>\IP Settings. The MAC Address is read-only.

To program a MiCollab Client MiNET Softphone:

MiCollab Client MiNET Softphones are programmed based on current Mitel IP phones. Refer to the *MiVoice Office 250 DB Programming Help* for details.

CREATING PHONES FROM CSV FILES

This section contains the following information:

- **Feature Description** below

- Creating a CSV File on **page 360**

- Creating IP Phones from CSV Files on **page 360**

- Creating Digital Telephones from CSV Files on **page 362**

- Troubleshooting for Creating Phones from CSV Files on **page 366**

FEATURE DESCRIPTION

A CSV file is an industry-standard format for text files containing data fields delimited by commas. DB Programming expects a phone information file to use either the TXT or CSV file extension and adhere to the other properties of a CSV file. Remember the following when creating a CSV file:

- Each line in the file represents information for a single phone. Information cannot be continued from one line to another.

- Data fields are delimited by commas.

- Only printable characters are considered part of a data field. Control characters within a data field are ignored.

- Each line contains the same number of data fields.

- If a comma is to be considered part of the data, that entire data field must be escaped by double-quotes at the beginning and end of the data field (for example, the data: Jones, Jim would be represented by the data field: "Jones, Jim").

- If a double-quote character is to be considered part of the data in a field, the entire field must be escaped by double-quotes at the beginning and end of the field and the double-quote character itself must be escaped by a preceding double quote (for example, the data: Sara "S" would be represented by the data field: "Sara ""S""").

- If no MAC addresses are read from the file, or if the MAC addresses contain invalid characters, a MAC address default of 00:00:00:00:00:00 (for v3.1 or later) or 08:00:0F:00:00:00 (v3.0 or earlier) is used for each IP phone imported. You must edit the MAC addresses in the dialog box.

- If no extensions are read from the file, the first available extension for a DB Programming phone is used for each phone to be imported. You must edit the extensions in the dialog box to resolve the conflicts.

- Regarding headers:

  - To use user-defined headers in a CSV file, the headers must be listed on the very first line. DB Programming reads the first line of the file and if no digits are identified, the first line is assumed to contain headers.

  - If the first line is identified as a header by DB Programming, DB Programming reads each field and tries to match the field with a pre-defined phone attribute. These include "Extension," "Username," "Description," and "MAC Address."

    To identify a data field as a pre-defined phone attribute, DB Programming sets all of the characters to lower-case, and then reads the field. **Table 61** shows how DB Programming identifies substrings.

Table 61: Substrings and Headers

SUBSTRING	HEADER
ext	Extension
user	Username
desc	Description
mac	MAC Address

Headers are used when parsing the remaining fields in the CSV file. The order of the headers corresponds to the order of the fields in the remaining entries of the file. As DB Programming parses each field, it uses the header order to determine which type of attribute it is reading. For example, if "Extension" was the first data field read, the first data field of each successive line in the file is considered a phone extension.

- If any digits are present in the first line of the file, DB Programming assumes there is no header in the file and uses its predefined order of fields: "Extension," "Description," "Username," "MAC Address" (MAC Address is only assumed if importing IP phones, otherwise, only three fields are expected per phone).

- Additional fields and headers not associated with the predefined phone attributes, "Extension," "Username," "Description," or "MAC Address" are ignored.

CREATING A CSV FILE

To create a CSV file:

1. Open the template file, `Phones.csv`, located in C:\Program Files\Mitel\5000\Templates (v3.1 or later) on the DB Programming computer.

2. Type the information that you want to import to DB Programming. The following is an example of phone information listed in a Microsoft Excel® spreadsheet.

3. Save the data as a CSV file. The following is an example of the file data after it is saved.

CREATING IP PHONES FROM CSV FILES

You can create IP phones from created CSV files (see **page 360**). For instructions to create digital telephones from CSV files, see **page 362**.

To create IP phones from a CSV file:

Before release 5.1, DB Programming did not identify the phone type in an export and all phones defaulted to the 52xx/53xx type when imported unless the type was changed manually. In release 5.1and later, a new Type field was added to the export and import csv files. So now the phone type is saved during an export and can be identified during an import.

Auto Populate Phone Type Based on MAC Address

Before release 5.1, if the phone type was not specified in the import dialog, the phone type defaulted to 52xx/53xx. In release 5.1and later, if the phone Type field is blank, it is automatically populated based on their MAC addresses as follows:

- 52xx/53xx phone MAC addresses begin with 08:00:0F.

- 86xx phone MAC addresses begin with 00:10:36.

- UCA phone MAC addresses begin with A1:21:00.

If the MAC address does not match any of these three categories, the phone type is set to IP Softphone. (This may also occur if a UCA phone entry has an invalid extension identified in the MAC address.)

If the MAC address and Phone Type fields are blank then the phone type is set to 52xx/53xx.

NOTE	You can also use the Configuration Wizard (beginning at the IP Device Setup dialog box) to import IP phones from a CSV file. See "Launching the Configuration Wizard" on **page 59** for instructions on how to launch the wizard. For complete information about the Configuration Wizard, refer to the *MiVoice Office 250 DB Programming Help*.

1. Select System – Device and Feature Codes – Phones – **Local** or **Remote**.

2. Right-click in the right pane, and then select **Create IP Phones from File**.

3. Click **Browse** to select the file containing the CSV file. After DB Programming reads the file, the information appears in the list control. The list control contains a Status column identifying the following information for imported extensions:

   - **New**, if the imported extension is new.

   - **Merge**, if the imported extension already exists, but with differences in information between the imported and existing extension.

   - **Error**, if the imported extension has some incorrect information and must be changed. If the imported extension does not have an associated phone type, then the Mitel IP Phone type displays.

   The Import button is disabled until all errors and Merge discrepancies are resolved.

4. Do one of the following:

   - To batch change the type of several phones, select the fields, and then right-click and select **Batch Change Type**. The Batch Change Type dialog box appears. Select the type for the phones, and then click **OK**.

   - To delete phones, select the phone that you want to delete, and then right-click and select **Delete**.

- To merge and update the imported information with the existing information:

  - Double-click, or right-click and select **View Merge Details**. The View Phone Merge Details dialog displays showing the current and new information. Fields with discrepancies are highlighted.

  - Click **OK**.

NOTE	If the description, username, first name, or last name fields are too long, they will be truncated and a tool tip will display with this information. The username field has a maximum of 10 characters, and all other fields have a maximum of 20 characters.

- To correct an error, hover the mouse over the entry with the error and a tool tip will display identifying the incorrect information. Correct the information as necessary directly in the offending field.

- To create Users automatically when phones are imported, select the phone, and then select **Yes** in the Create User field.

5. Click **Import** to program the phones in the database. At any point, click **Cancel** or click on the red "X" at the top of the dialog box to cancel the creation of new phones and return to the prior view of DB Programming.

CREATING DIGITAL TELEPHONES FROM CSV FILES

To create digital telephones from a file:

1. Select System – Controller – **Digital Expansion Interface**.

2. Double-click the Digital Expansion Interface that you are using, and then click **Digital Endpoint Module - 16**.

3. Right-click the port(s), and then select **Create Digital Telephones from File**.

   The Create Phones From File dialog box appears.

4. Click **Browse** to select the file containing the information for the phones. After DB Programming reads the file, the information appears in the list control.
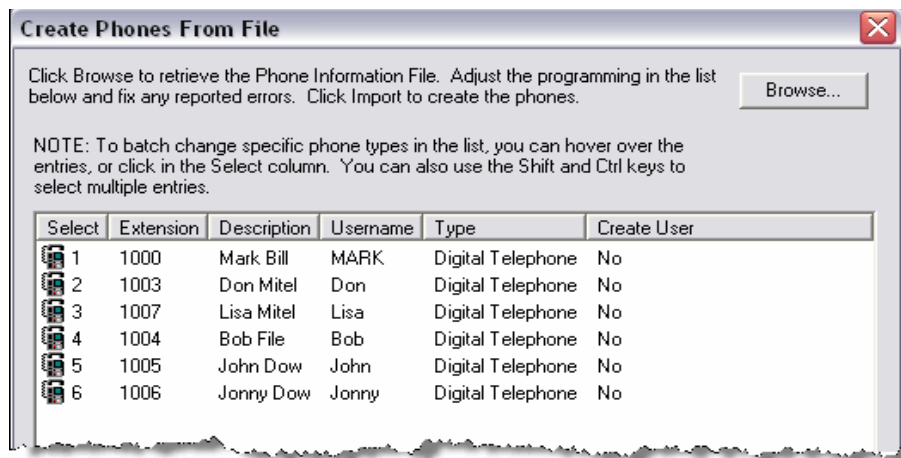
The Type fields are listed as "Digital Telephones" because only digital telephones may be programmed in the DEM-16 ports through this dialog box.

To delete phones, select the phone that you want to delete, and then right-click and select **Delete**.

DB Programming validates the information and reports any issues. If there are no errors or conflicts, proceed to step **5**. If any errors exist, the Import button is disabled. See **page 366** for a list of possible error messages and troubleshooting information.

Edit the information in the list and after all entries are valid, the Import button becomes enabled. Then proceed to step **5**.

5. Click **Import** to program the phones in the database. At any point, click **Cancel** or click on the red "X" at the top of the dialog box to cancel the creation of new phones and return to the prior view of DB Programming.

USING PIN ACTIVATION FOR 52XX/53XX PHONES

You can use a Personal Identification Number (PIN) to register 52xx/53xx phones to the MiVoice Office 250. You no longer have to assign MAC addresses for 52xx/53xx phones manually. This PIN Registration feature applies to offline 52xx/53xx phones only and is enabled by default. You can also enable the PIN Replacement feature to replace the MAC address of a programmed 52xx/53xx phone with the MAC address of an unprogrammed 52xx/53xx phone.

Version 4.0 or later software allows Mitel IP phone activation using a Personal Identification Number (PIN) instead of using a preconfigured Media Access Control (MAC) address. This saves time during installation by letting administrators associate a physical phone with a logical extension from the phone itself. (If desired, administrators can continue to pre-assign MAC addresses manually.) Supported IP phones include the 5304, 5212, 5224, 5312, 5324, 5330, 5340, and 5360.

The "PIN Registration" option allows system administrators to program a block of "free" IP phone extensions that are not yet associated with any particular MAC addresses. The free extensions appear as Mitel 52xx/53xx IP phones with no MAC addresses. When an IP phone with an unrecognized MAC address connects to the MiVoice Office 250, the phone's display prompts the administrator for a PIN. The administrator enters the PIN and then presses the Hold button to send the PIN to the system for validation. (For example, to register extension 1000 using access code 1234, enter 12341000 and then press the Hold button.) If the PIN is rejected, the phone continuously prompts for a new PIN. If the PIN is accepted, the system permanently assigns the phone's MAC address to the extension number that matches the PIN.

The "PIN Replacement" option allows system administrators to replace the MAC address of a programmed Mitel IP phone with the MAC address of an unprogrammed Mitel IP phone. This feature is useful for replacing an existing programmed phone without having to first enter the MAC address of the replacement phone in DB programming. (The PIN Registration option will not work in this case because the extension already has an assigned MAC address.)

In v4.0 or later software, PIN Registration is enabled by default and PIN Replacement is disabled.

PIN Registration and PIN Replacement apply only to offline extensions and only to supported Mitel IP phones. If the administrator enters a PIN that matches the extension of an online phone, the system will reject the PIN and prompt for another PIN.

When both PIN Registration and PIN Replacement are enabled, the rules of both individual settings apply. If neither PIN Registration nor PIN Replacement is enabled, the system will not allow a phone to prompt for a PIN, and the connection will be rejected just as it would be with earlier versions of software.

The system allows ten PIN entry attempts before locking out the phone. When a phone is locked out, it must be reset before the system will allow it to prompt for a PIN again.

This section contains the following information:

- PIN Registration and PIN Replacement on **page 364**
- Enabling PIN Registration Per Phone on **page 365**
- Resetting MAC Addresses on **page 365**
- Refreshing MAC Addresses on **page 368**

PIN Registration and PIN Replacement

The following fields are added to the System\Phone-Related Information folder support the PIN Registration and PIN Replacement features for 52xx/53xx phones:

- **Enable PIN Registration**: Enables/disables registration by PIN. It is set to Yes by default.

- **Access Code for PIN Registration**: If PIN registration is enabled, this code is used with the extension to send the PIN to the system when a 52xx/53xx phone comes online. (For example, to register extension 1000 using access code 1234, enter 12341000 and then press the Hold button.) If registration is disabled, this field appears with a red "X."

- **Enable PIN Replacement**: Enables/disables replacement by PIN. It is set to No by default.

- **Access Code for PIN Replacement**: If replacement is enabled, this code is used with the extension to send the PIN to the system when a 52xx/53xx phone comes online. (For example, to register extension 1000 using access code 1234, enter 12341000 and then press the Hold button.) If replacement is disabled, this field appears with a red "X."

To program the access code for PIN Registration:

1. Select System – Phone-Related Information – **Access Code for PIN Registration**.

2. Type the access code in the text box. The range is 0-11 characters (digits, *, or #).

3. Click out of the field or press **Enter** to save the change.

To disable the PIN Registration feature for all 53xx/53xx phones:

1. Select System – **Phone-Related Information**.

2. Change the Enable PIN Registration flag to **No**. The Use PIN Registration check box in the Create 52xx/53xx Extension dialog box for 52xx/53xx phones is cleared as well.

To enable the PIN Replacement feature for all 53xx/53xx phones:

1. Select System – **Phone-Related Information**.

2. Change the Enable PIN Replacement flag to **Yes**.

To program the access code for PIN Replacement:

1. Select System – Phone-Related Information – **Access Code for PIN Replacement**.

2. Type the access code in the text box. The range is 0-11 characters (digits, *, or #).

3. Click out of the field or press **Enter** to save the change.

Enabling PIN Registration Per Phone

There is a new check box named Use PIN Registration in the Create 52xx/53xx Extension dialog box.

To enable the PIN Registration feature for a phone:

1. Select System – Devices and Feature Codes – **Phones**.

2. Right-click anywhere in the right pane, and then select **Create 52xx/53xx**. The Create 52xx/53xx Extension dialog box appears. See step 3 on **page 356** for a example of the Create 52xx/53xx Extension dialog box.

3. Make sure the **Use PIN Registration** check box is selected. This is enabled by default. The MAC Address section displays "Registered by PIN" instead of a programmable MAC Address.

4. Click **OK**. The MAC address field in System\Devices and Feature Codes\Phones\<*IP phone*>\IP Settings\Network Configuration is programmed with "00:00:00:00:00:00." The Enable PIN Registration flag in the Phone-Related Information folder remains Yes.

To disable the PIN Registration feature for a phone:

1. In the Create 52xx/53xx Extension dialog box, clear the **Use PIN Registration** check box. Note that the Enable PIN Registration flag in the Phone-Related Information folder is not changed to No because other phones already programmed may still be registered by PIN.

2. Click **OK**.

Resetting MAC Addresses

You can reset the MAC addresses for 52xx/53xx phones.

To reset the MAC address for a 52xx/53xx phone:

1. Select System – Devices and Feature Codes – Phones – <*52xx/53xx phone*> – IP Settings – **Network Configuration**.

2. Right-click the MAC Address, and then select **Clear**. The MAC address is reset to "00:00:00:00:00:00."

To reset the MAC addresses for multiple 52xx/53xx phones:

1. Select System – Devices and Feature Codes – **Phones**.

2. Select one or more 52xx/53xx phones, and then right-click and select **Clear PIN Registration**. The MAC address(es) are reset to "00:00:00:00:00:00."

Entering the PIN on a Phone

To enter the PIN on a phone, dial the assigned access code (see **page 364**) and the phone's extension number, and the press the Hold button. For example, to register extension 1000 using access code 1234, enter 12341000 and then press the Hold button.

TROUBLESHOOTING FOR CREATING PHONES FROM CSV FILES

Table 62 lists troubleshooting information for importing phones from CSV files.

Table 62:   Create Phones from File Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION	IP PHONE	DIGITAL TELEPHONE
One or more extensions are invalid.	One or more extensions may: • conflict with existing extensions in the database. • conflict with any extensions in the dialog box. • contain invalid characters. • exceed five digits.	Click any of the extension fields, and then either type in a new extension or select a valid extension from the Extension list.	✓	✓
One or more descriptions contain invalid characters.	Descriptions are restricted to 20 characters and all printable characters except for "|."	Click the description in the list, and then remove the invalid characters from the field.	✓	✓
One or more usernames contain invalid characters.	User names are restricted to 10 characters and all printable characters except for "\," "~," and "|."	Click the user name in the list, and then remove the invalid characters from the field.	✓	✓
One or more usernames were too long and have been truncated.	If more than 10 characters exist for any user name imported, the user name is truncated to 10 characters	None. As designed.[1]	✓	✓
One or more descriptions were too long and have been truncated.	If more than 20 characters exist for any description imported, the description is truncated to 20 characters.	None. As designed.[1]	✓	✓
One or more MAC Addresses are invalid.	One or more MAC Addresses may: • conflict with existing MAC Addresses in the database. • conflict with any MAC Addresses in the dialog box. • contain invalid characters,.	Click any of the MAC Address fields, and then edit the field in the Edit MAC Address box.	✓	

Page 1 of 2

Table 62:   Create Phones from File Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION	IP PHONE	DIGITAL TELEPHONE
You may only import X IP Phone(s). You will need to delete Y phone(s) to continue with the import.	The number of phones read from the file exceeds the system limit of 250.	Delete Y phones from the list.	✓	
You selected X port(s) for installing new phones. You will need to delete Y phone(s) to continue with the import.	More phones are read from the file than ports selected. The X indicates the number of ports selected before invoking this dialog and the Y indicates the number of phones to delete from the phones listed.	Delete Y phones from the list.		✓

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1. This error message does not disable the Import button.

CREATING OFF-NODE DEVICES

Off-node devices give node users access to devices (for example, other system extensions) on other nodes. Each off-node device is identified with an extension number, description, username, device type (phone, hunt group, application, or page port), and node identification number. The description and username options are for directory features only. For more information about system nodes, see "Private Networking and System Nodes" on **page 91**.

> **NOTICE**
>
> **Possible System Instability**. Do not create or delete more than 2000 off-node devices at a time. Batch creating more than 2000 off-node devices may cause problems with the system.

Only the local directory features are affected when you change the description and username of an off-node device—remote nodes do **not** send updates to other nodes when off-node device information is changed. Only the local node for that device sends out any updates. For example, if the device is programmed as John Doe on node 1, but you change the associated off-node device to Jane Smith on node 2, the node 2 IC directory reflects Jane Smith. The device on node 1, however, still displays John Doe.

You can create off-node devices for phones on the other nodes and program individual phones on the Local node.

To select the node to program:

1.   Select System – Devices and Feature Codes – **Nodes**.

2.   Double-click the node number. The current list of off-node devices for that node appears.

To create off-node devices:

1.   Select System – Devices and Feature Codes – **Phones**.

2.   Right-click anywhere in the right pane to view the list of options, as shown below.
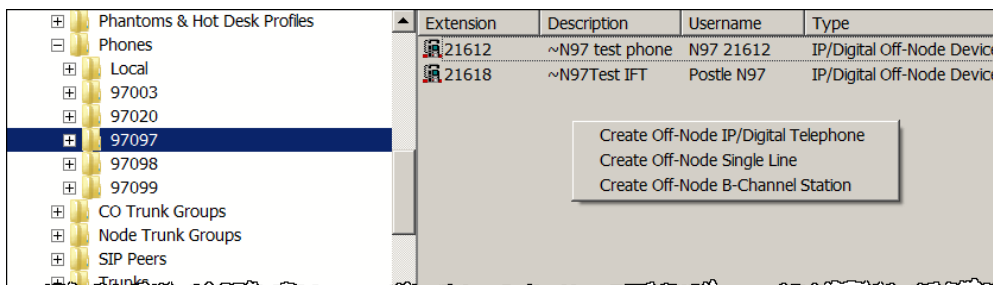
3. Select one of the following options:
    - Create Off-Node IP/Digital Telephone
    - Create Off-Node Single-Line
    - Create Off-Node B-Channel Station

4. Depending on your selection, the Create *<off-node device>* dialog box appears.

5. Select the starting extension number or enter the wildcard extension for the devices. For more information about wildcard extensions, see the following section, "Using the Wildcard Character in Off-Node Extensions" on **page 368**.

6. If applicable, enter the number of extensions.

7. Click **OK**. If you entered the same number as an existing extension, an error message appears, and you must enter a new number. The new off-node device appears in the list without a description or username.

USING THE WILDCARD CHARACTER IN OFF-NODE EXTENSIONS

Wildcard extensions can only be used for off-node devices. You can use the wildcard character X in extension numbers to represent "any digit." This allows you to include a range of extensions as one off-node device entry.

Wildcard extensions are made up of digits (1–9), followed by wildcard digit X. Examples of valid wildcard extensions are 1XXX (range of 1000–1999), 14XX (range of 1400–1499), 7X (range of 70–79). For example, if there is a 14XX wildcard, users can dial 1433 and be connected to that off-node device.

The following limitations apply to using wildcard extensions:

- There is no entry for wildcard extensions in the intercom directory. Direct Station Select/ Programmable Key Module (DSS/PKM) buttons can be programmed for the individual devices the wildcard extension is representing.

- Voice Processing cannot create associated mailboxes for phones included in wildcard ranges. To have an associated mailbox, the off-node phone must have its own off-node device entry on the node where the external voice processing system is located.

Refreshing MAC Addresses

Whenever the Network Configuration folder for a 52xx/53xx phone is refreshed, DB Programming refreshes the MAC Address field. You can manually refresh the MAC address for an 52xx/53xx phone if the connection is in online mode.

To manually refresh the MAC address for an 52xx/53xx phone:

1. Select System – Devices and Feature Codes – Phones – *<52xx/53xx phone>* – IP Settings – **Network Configuration**.

2. Right-click the MAC Address, and then select **Refresh**. The MAC address field is refreshed. This verifies whether or not the phone has been registered yet and the MAC Address has been automatically programmed.
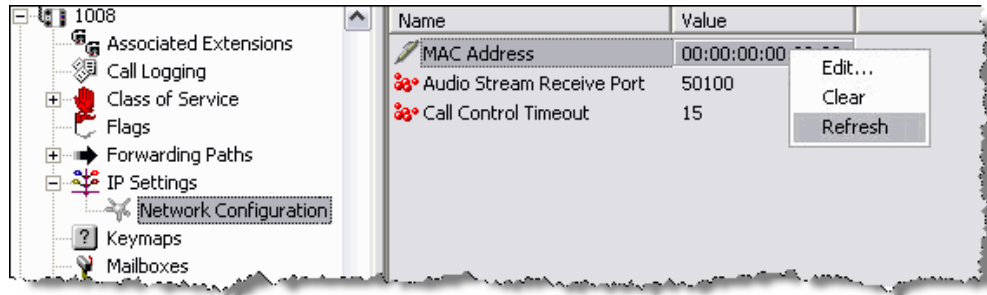
PROGRAMMING THE DIGITAL EXPANSION INTERFACE (DEI)

Bay designations appear in the Controller folder, displaying which modules are installed in the specific bay locations for the main chassis and DEI(s).

Indicates bays in which modules are installed
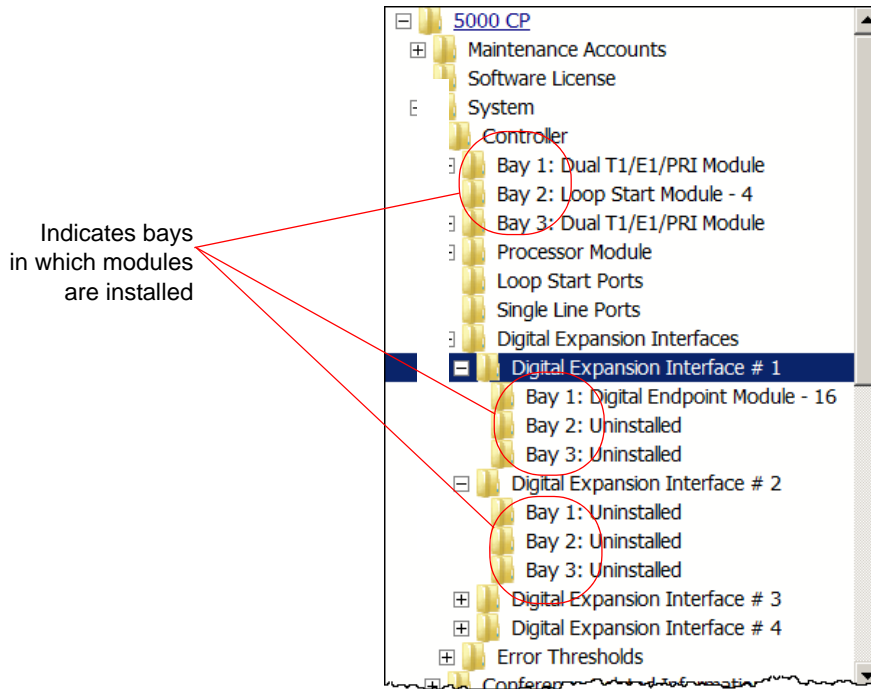
PCDPM Configuration

After the PCDPM hardware is installed, the telephone system database must be programmed for the PCDPM intended purpose. For PCDPM installation instructions, refer to the *MiVoice Office 250 Installation Manual* .

To configure DB Programming for PCDPM use:

1. Start DB Programming, then navigate to the System\Controller folder and select the Digital Expansion Interface where the PCDPM is installed.

2. Set the following options:

- **System Configuration:** In the Digital Endpoint Module - 16 window for the DEI where this PCDPM will be used, select the Port Type for the port assigned to the PCDPM.
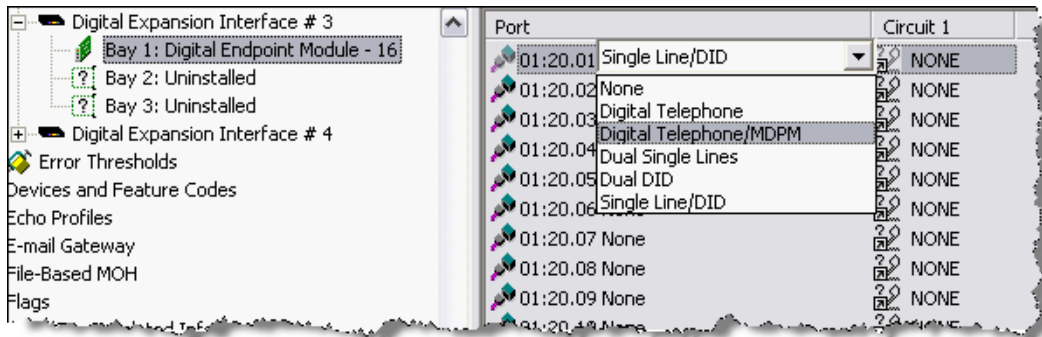
- Select **Digital Telephone** if you plan on using the PCDPM as a PKM (DSS/BLF) unit connection or for Off-Hook Voice Announce receive capability.

- Select **Digital Telephone/MDPM** if attaching an MDPM to the PCDPM and you still want to use the PCDPM for Off-Hook Voice Announce receive capability. The serial connection on the PCDPM is *not* functional; therefore, Desktop Interface through serial connections is not supported. Only the PKM 16 (Mini-DSS) is supported, and it does not require a PCDPM.

PROGRAMMING A DIGITAL DESKTOP MODULE - 16 (DDM-16)

When the System is an HX Controller, you can program up to three DDM-16s. The modules allow for 48 additional voice paths, beyond the 192 supported previously in 4-DEI Mode, allowing for a total of 240. Note that 2-DEI and 4-DEI Modes are unchanged and because licensing is handled on a per-phone basis, licensing is not an issue for DDM-16s.

If you attempt to create more than three DDM-16s, an error message appears indicating that the fourth DDM-16 is not operational.

With the introduction of the HX Controller chassis, DB Programming makes all attempts to appropriately show the hardware that is available for both HX and CS systems. If a DDM-16 is programmed into a database and then the platform is changed to CS Controller, a red X appears over this module.

Programming a DDM-16:

When the System is an HX Controller, the new module type, DDM-16, appears as a choice in the drop-down list when one of the Bays is clicked in the Controller folder.

The programming of devices on the new DDM-16 is identical to the programming of devices on the existing DEM-16 (see **page 370**).

You can also program a DDM-16, using the Configuration Wizard. See "Launching the Configuration Wizard" on **page 59** for instructions on how to launch the wizard. For complete information about the Configuration Wizard, refer to the *MiVoice Office 250 DB Programming Help*.

Swapping Modules

You can move a module between Bays 1-4 in offline mode.

To swap modules:

1. Select the module to move, and then do one of the following:

    • Drag and drop the module to the new location.

    • Press **CTRL** + the up/down arrow to move the module up or down in the list.

2. When the confirmation message appears, click **Yes**.

CONFIGURING A FOUR-PORT SINGLE LINE MODULE (SLM-4)

Single Line ports on the SLM-4, and the two on-board Single Line ports, may be configured with single lines phones. The two Single Line ports and SLM-4 can also be configured using the Configuration Wizard. See "Launching the Configuration Wizard" on **page 59** for instructions on how to launch the wizard. For complete information about the Configuration Wizard, refer to the *MiVoice Office 250 DB Programming Help*.

When you double-click on a Single Line Module -4, a list of the circuits appears on the right of the screen. This allows you to configure the SLM-4 ports to support up to four Single Line devices.

NOTE	If you attempt to configure more than one SLM-4 board in the CS Controller, the system refuses the change and displays an error message. To support multiple SLM-4s (up to four), use the HX Controller.

When the System is an HX Controller, the on-board Single Line port capacity is expanded from two to four ports. DB Programming shows four onboard Single Line ports. Although the boards are set up as 4-port boards, only the original two ports are equipped by default. The figure below shows an example of the Singe Line Ports 1-4.
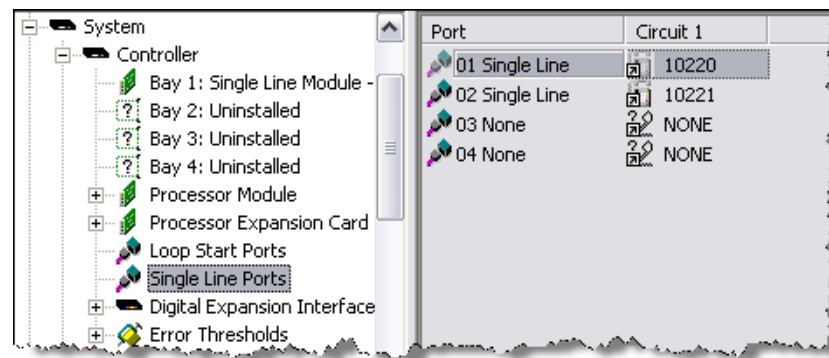
With the introduction of the HX Controller chassis, DB Programming makes all attempts to appropriately show the hardware that is available for both HX and CS systems. If three Single Line ports are programmed into a database and then the platform is changed to CS Controller, a red X appears over the unavailable third port.

To equip a single port:

1. Select the current Port description and use the drop-down list box to scroll to the new port type.

2. Press **ENTER** or click another field to save the change.

To equip several ports at once:

1. Select the ports you want to equip.

   To select a series of items, hold down **SHIFT** while selecting the first and last item in the range. To select two or more that are not consecutive, hold down **CTRL** while selecting the desired items.

2. Right-click and select **Batch Change Type**. The Batch Change Type dialog box appears.

3. Click the port type you want to assign, and then click **OK**.

If the new devices require extension numbers, you will be prompted to select the first number in the batch. The available extension numbers will be assigned sequentially to the newly equipped ports. If you attempt to exceed the device limit of the System, an error message appears informing you the operation cannot be performed.

Moving a Single Line Port

When DB Programming is in offline mode, you can drag/drop a port to a module that has available ports ("None"). You cannot move devices in online mode. If you try to drop on a fully programmed module, an error message appears.

To "move" a Single Line port:

1. Open the desired database in an offline mode session of DB Programming. Make sure the desired module is explored so the "None" port move location is viewable in the left-hand tree view.

2. Browse to the module with the existing port you want to move. Click the port, hold down the mouse button, and drag it over to the tree view. The None port will become highlighted when the port to be moved is dragged onto it.

POWER FAIL BYPASS CAPABILITIES

The HX controller has four loop start and four single line ports (the SL port addresses are 11.1.1, 11.2.1, 11.3.1, 11.4.1). The first two ports are used for the Power Fail Bypass feature. This feature allows up to two single line phones to be able to make calls after a total system shutdown due to power loss. The CS controller provides two loop start ports and two single line ports (the port addresses are 11.1.1 and 11.2.1). These ports are used for the Power Fail Bypass feature. Refer to the MiVoice Office 250 Installation Manual for information on how the ports are cabled.

TROUBLESHOOTING THE FOUR-PORT SINGLE LINE MODULE

Table 63 summarizes the troubleshooting strategies recommended for resolving SLM-4 discrepancies.

Table 63: Single Line Module (SLM-4) Troubleshooting Tips

Symptom	Possible Cause	Corrective Action
SLM-4 does not work after being plugged in.	Software prior to v2.x does not support SLM-4.	Upgrade all software to 2.x using the upgrade utility.

Table 63:   Single Line Module (SLM-4) Troubleshooting Tips(continued)

Symptom	Possible Cause	Corrective Action
SLM-4 does not work when plugged in.	SLM-4 is not programmed in DB Programming.	Mitel does not support Auto-Equip for SLM-4 modules. When the module is plugged in for the first time, it still has to be programmed in DB Programming for the appropriate bay.
After plugging in multiple SLM-4 modules only one is working.	Only one SLM-4 can be installed in the CS Controller.	To install multiple SLM-4s, use the HX Controller. The HX Controller supports up to four SLM-4s.
DTMF receiver not available for SLM-4 or the SL module.	All of the DTMF receiver resources are in use.	Although this occurrence is extremely rare, try again when resources free up.

PROGRAMMING AN EIGHT-PORT SINGLE LINE MODULE

Up to three eight-port single line modules (SLM-8) can be installed in a Digital Expansion Interface (DEI) chassis, and up to eight single line phones may be configured with each SLM-8.

You can program SLM-8 modules with the Configuration Wizard or directly in DB Programming. See "Launching the Configuration Wizard" on **page 59** for instructions on how to launch the wizard. For complete information about the Configuration Wizard, refer to the *MiVoice Office 250 DB Programming Help*.

The direct process appears in the following sections. For quick-reference to details while programming, refer to the online Help.

PROGRAMMING OVERVIEW

To activate and manage an SLM-8 module, perform the following procedures as needed.

If you do not use the Configuration Wizard to set up an SLM-8 module, carry out the procedures in the following process in DB Programming. Step-by-step procedures appear in the "Programming Procedures" section next.

1. Program an SLM-8 module in a DEI bay.

2. Set up one single line port with a system circuit number, as appropriate.

3. Set up multiple, or batch, single line ports with system circuit numbers, as appropriate.

4. Program individual phones in accordance with customer requirements.

Once an SLM-8 module has been installed and is in operation, the following procedures can be used to make changes to, or remove, existing assignments. Procedures for the following tasks also appear in the section titled "Programming Procedures" next:

• Change a phone circuit number.

• Remove a single line circuit number.

• Remove an SLM-8 module from system programming.

You can also configure the SLM-8 module with the Configuration Wizard. For further information about this option, see **page 376**.

PROGRAMMING PROCEDURES

The following procedures apply to the SLM-8 module.

To add an SLM-8 module to a DEI chassis:

1. Start the MiVoice Office 250 Session Manager. The DB Studio window appears.

2. From the left pane of the DB Studio window, select System – Controller – **Digital Expansion Interface <*DEI n*>**. Status of Bay 1, Bay 2, and Bay 3 appears in the right pane.

3. Click an Uninstalled bay. A drop-down arrow appears in the box.

4. Click the arrow and select **Single Line Module - 8** from the drop-down list.

5. Click anywhere in the right pane. A pop-up window appears asking if you're sure you want to make the change.

6. Click **Yes**.

To set up one SLM-8 single line port and circuit number:

1. Make sure an SLM-8 module is installed in a system DEI.

2. Start the MiVoice Office 250 Session Manager. The DB Studio window appears.

3. From the left pane of the DB Studio window, select System – Controller – **Digital Expansion Interface <*DEI #: Single Line Module - 8*>**. When you click the desired Bay n: Single Line Module - 8, the status of the SLM-8 ports appears in the right pane.

4. Click the Port icon to the left of the port number you want to use. A drop-down box and arrow appear.

5. Click the drop-down arrow and select **Single Line**.

6. Click anywhere in the right pane. The Create Single Line Extension window appears.

7. Click the drop-down arrow of the Starting Extension box. A drop-down list of system circuit numbers unfolds.

8. Scroll to the unused circuit number you want to assign, select it, and click **OK**. The selected system circuit number, or telephone extension, appears in the Circuit 1 column next to the Port you chose.

To set up multiple, or batch, SLM-8 single line ports and circuit numbers:

1. Make sure an SLM-8 module is programmed in a system DEI.

2. Start the MiVoice Office 250 Session Manager. The DB Studio window appears.

3. From the left pane of the DB Studio window, select System – Controller – **Digital Expansion Interface <*DEI #: Single Line Module - 8*>**. When you click the desired Bay n: Single Line Module - 8, the status of the SLM-8 ports appears in the right pane.

4. Click the Port icon to the left of the first port number you want to use. The port number field changes color and a drop-down box and arrow appear.

5. If the ports you want to program are in sequence, to select the batch of ports, press **Shift** + the final circuit in the batch you want. The selected port number fields change color. If the ports are not sequential, hold down **Ctrl** and then select the desired port numbers one at a time. The selected port number fields change color.

6. Right-click the top selected port number. The Batch Change Type... command button appears, as shown in the following illustration.
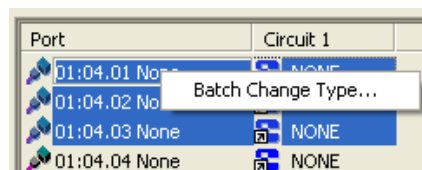
7. Click Batch Change Type. The Batch Change Type window appears (not shown).

8. Select the **Single Line** option button and click **OK**. The Create Single Line Extension window appears.

9. Click the drop-down arrow, scroll to and select the Starting Extension from the list of extension numbers, and click **OK**. Available sequential circuit numbers are assigned to the selected ports and appear in the Circuit 1 column, where **NONE** appears prior to assigning circuit, or extension, numbers.

To program single line phones:

1. Start the MiVoice Office 250 Session Manager. The DB Studio window appears.

2. From the left pane of the DB Studio window, select System – Devices and Feature Codes – Phones – ***<local extension number>***.

3. Program the single line phone in accordance with user requirements.

To uninstall an SLM-8 module:

1. Start the MiVoice Office 250 Session Manager. The DB Studio window appears.

2. From the left pane of the DB Studio window, select System – Controller – Digital Expansion Interface *<DEI n>* – **Single Line Module - 8**.

3. Click the Bay icon to the left of the module name. The drop-down arrow appears.

4. Click the drop-down arrow and select **Uninstalled**.

PROGRAMMING DEVICE DESCRIPTIONS AND USER NAMES

Device descriptions are used in the IC directory and user names appear on phone displays. Also, because the Intelligent Directory Search (IDS) feature (see the following section) searches for phone descriptions and user names, the information entered in the phone Description and Username fields affects how people search for information in the system.

For more information about Directory, see "Directory of Intercom, Speed Dial, and Feature Codes" on **page 290**.

NOTE	If you do not program the description and username for the phones on the local node, the local phones will not display in the IC directory. Also, only the local directory features are affected when you change the Username and Description of an off-node device. Remote nodes do not send updates to other nodes when off-node device information is changed. Only the local node for that device sends out updates. For example, if the device is programmed as John Doe on node 1, but you change the associated off-node device to Jane Smith on node 2, the node 2 IC directory reflects Jane Smith. The device on node 1, however, still displays John Doe.

To enter a a device description or username:

1. Select System – Devices and Feature Codes – **Phones**.

2. In the **Extension** column, select **Description** or **Username**, and then enter the new information in the box. Descriptions can contain up to 20 characters and usernames can contain up to 10 characters. To ensure proper operation of voice processor directories, enter the full name in the form: Last name, First name. Do not use slash (/), backslash (\), vertical bar (|), or tilde (~) characters in user names. Do not use Control characters or punctuation marks in descriptions or usernames.

3. Click out of the field or press **ENTER** to save your change.

IDS SUPPORT

The MiVoice Office 250 supports Intelligent Directory Search (IDS), which is similar to the "text on 9 keys" (T9) predictive search feature used for mobile phones. For more information about IDS, see "Intelligent Directory Search" on **page 290**.

HIDING USER NAMES IN VOICE MAIL DIRECTORIES

You can hide programmed descriptions in the Voice Mail Directory. For example, a user may have a primary extension at the office programmed as "Bond, James." However, the same user may have a secondary extension—for example, at home—that is not published from the Voice Mail Directory.

Because the IC directory is intended for internal use, both primary and secondary extension entries appear in the IC directory. There may be any number of primary (no tilde in description) and secondary extensions (first character of description is a tilde). To differentiate between these two, an asterisk appears immediately before the secondary extension entry—for example, "*James Bond".

To hide an entry in the Voice Mail directory:

Enter a tilde (~) in the Description field immediately before the last name—for example, "~BOND, JAMES". The tilde character (~) makes the entry inaccessible in the Voice Mail directory. Note that if someone knew the number of the secondary extension, they could dial that extension directly. Someone entering "BOND" (2 6 6 3) at the Voice Mail prompt, however, would find only the primary extension.

COPYING PHONE PROGRAMMING

If you have multiple phones that use the same programming, you can copy the programming from one phone to the others.

To copy a phone:

1. Select – System – Devices and Feature Codes – Phones – <***Local***>.

2. Right-click the extension number that you want to copy, and then select **Copy**.

3. Right-click the phone in which you want the copied settings pasted, and then select **Paste**. A dialog box appears, similar to the one shown in the following example.
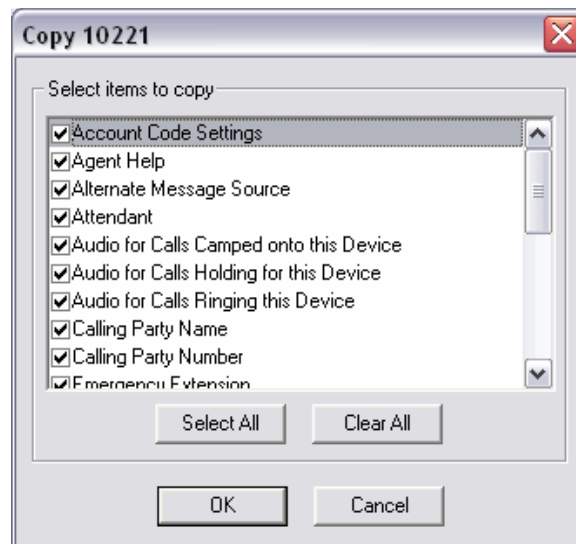
4. Select or clear the attributes that you want to copy to the phone.

5. Click **OK** to save the changes.

VIEWING ASSOCIATED DEVICES AND REFERENCES

You can use the Associated Devices and References feature to view the associated devices for a phone, mailbox, or hunt group. You can also use the Associated Devices and References feature to query various groups in the database to locate the associated phone to the extension.

To view Associated Devices:

1. Select – System – Devices and Feature Codes – **Phones**.

2. Right-click the phone, and then select **Associated Devices and References**. A dialog box, similar to the one below, appears.
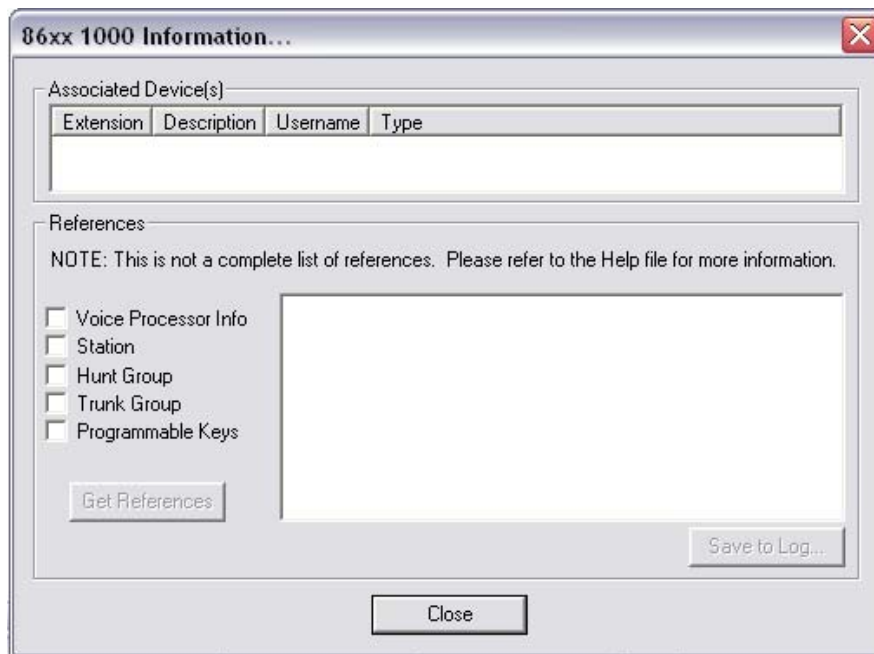
CHANGING PHONE EXTENSION NUMBERS

You can change phone extension numbers.

NOTE	When changing user names or extensions, 5320, 5330, 5340, and 5360 self-labeling programmable buttons may need to be refreshed to display the new data. To reduce the load on the system, the self-labeling buttons do not refresh until 30 seconds after database changes have occurred.

CHANGING A SINGLE EXTENSION NUMBER

You can change phone extension numbers.

To change the phone extension number:

1. Select System – Devices and Feature Codes – **Phones**.

2. In the **Extension** column, select the current extension number that you want to change.

3. Select an available extension from the list, or enter the new number in the box. If you change the phone extension to match a previously existing unassociated mailbox, a warning window appears informing you that the new extension is the same as an unassociated voice mailbox.

4. Click **OK** to associate the mailbox with the current device.

CHANGING MULTIPLE EXTENSION NUMBERS AT ONE TIME

You can use the Batch Extension Change feature to change multiple extension numbers at one time.

To use Batch Extension Exchange:

1. Select System – Devices and Feature Codes – **Phones**.

2. In the **Extension** column, select the phones you want to change (you can use the SHIFT or CTRL key to select more than one phone), right-click, and then select **Batch Extension Change**. A dialog box similar to the following appears.

3. Select the number that you want to assign to the first selected phone; the other phones will be numbered consecutively after this number.

4. Click **OK**. The phones are automatically renumbered and resorted in the phone list.

CLEARING PIN REGISTRATION

See **page 365** for details.

CONVERTING USERNAMES TO MIXED CASE

Previous versions or DB Programming allowed only uppercase letters for user names. You can convert uppercase user names to mixed case. When you convert user names to mixed case, DB Programming selects the first letter of the user name to be uppercase and converts the rest of the letters to lowercase. In addition to the first letter of a user name, the mixed case conversion also converts letters following a space or apostrophe to uppercase (for example, "o'neil, abc" is converted to "O'Neil, Abc").

NOTE	When changing user names or extensions, 5320, 5330, 5340, and 5360 self-labeling programmable buttons may need to be refreshed to display the new data. To reduce the load on the system, the self-labeling buttons do not refresh until 30 seconds after database changes have occurred.

To convert a user name:

1. Select System – Devices and Feature Codes – Phones – <***Local***>.

2. In the **Username** column, right-click the user name that you want to convert (for example, DOW, JOHN).

3. Select **Convert Usernames to Mixed Case** from the list. The user name is converted to mixed case, for example, *Dow, John*, as shown in the following example.
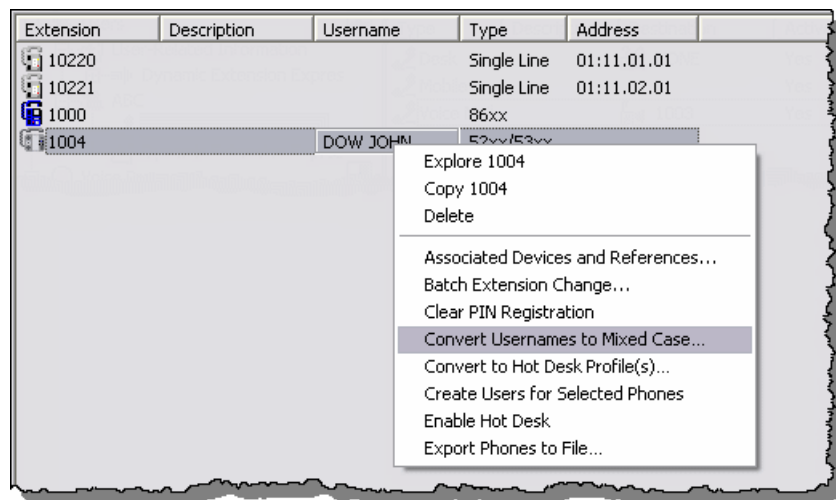
After converting

CONVERTING TO 52XX\53XX PHONES

If you have replaced 86xx phones with 52xx\53xx phones, you can convert the settings in DB Programming. The following are default settings for converted phones:

- Audio Stream Receive Port (defaults to 50100)

- NAT Address Type (defaults to Native)

- Standard/Alternate Keymap Group (defaults to Default Keymap)

This is available only in offline mode. This option is located in System\Devices and Feature Codes\Phones.

To convert 86xx phones to 52xx\53xx phones:

1. In an offline mode session, select the extension(s) that you want to convert.

2. Right-click and select **Convert to 52xx/53xx…**. The Convert Digital Phones to 52xx/53xxIP Phones dialog box appears.
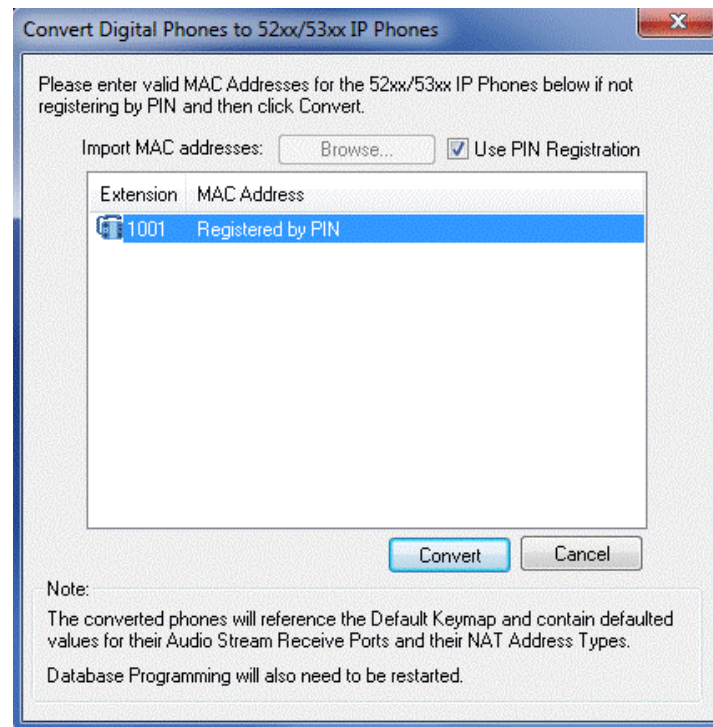
3. Register the phone by programming a MAC address or using a PIN (see **page 386** for details):

| NOTE | If the Digital telephone has an associated MDPM (Modem Data Port Module) attached to it, then you will be asked if you want to delete the MDPM. If you delete the MDPM, the digital phone will be converted to a 52xx/53xx IP Phone and the attached MDPM extension will be deleted. If you do not delete the attached MDPM, then the conversion will be cancelled. |

4. After all of the MAC addresses are valid, or you are registering the phone using a PIN, click **Convert** to initiate the conversion process. A message displays to restart DB Programming.

> **NOTE** | At any point before clicking **Convert**, you may cancel the conversion by clicking **Cancel**.

5. Restart DB Programming.

EDITING A SINGLE MAC ADDRESS

All the phones in the list must have valid MAC addresses programmed. These MAC addresses must not conflict with one another or with any MAC addresses of devices programmed in the database. If the MAC addresses are invalid, a red "X" appears.

To edit a single MAC address:

1. Click the MAC address that you want to program (from step 3 on **page 385** above. The Edit MAC Address dialog box appears.

2. Type the new MAC address, and then click **OK**.

EDITING MULTIPLE MAC ADDRESSES

You can edit multiple MAC addresses at one time.

To program batch MAC addresses:

1. From the dialog box that shows the MAC address (from step 3 above on **page 385**), click **Browse** to import the MAC addresses from a `.txt` file. The standard Windows Open dialog box opens. When you import the file, the MAC addresses are read from the top. This means that if there are six MAC addresses in the file, but there are eight phones, only the first six will be assigned an address. Also, if there are eight MAC addresses in the file, but only six phones, only the first six addresses will be used. All MAC addresses must be unique.

2. Click **OK** to accept the extensions and MAC addresses.

CONVERTING TO HOT DESK PROFILE(S)

You can convert any phones that support Hot Desking to Hot Desk Profiles. The supported phones are Digital Telephones, 86xx IP phones, 52xx/53xx IP phones, and 8602 Softphones. Converting a phone to a Hot Desk Profile retains all of the programming for that phone, such as programmable keys, speed dial numbers, etc. See **page 447** for details about Hot Desking.

NOTE	The Convert to Hot Desk Profile(s) option is available in offline mode only. Converting phones to Hot Desk Profiles requires a system reset.

To convert a phone to a Hot Desk Profile:

1. Select System – Devices and Feature Codes – **Phones**.

2. Right-click the phone you want to convert, and then select **Convert to Hot Desk Profile(s)**.

3. When the system prompts you whether you want to keep the phone, do one of the following:

    - Click **Yes** to change the phone extension and keep the phone. Keeping the phone allows the physical phone to stay in place and be used as a "hot desk base" that the user can log into. However, you must change that phone's extension so that the original extension can be used for hot desking. (The Hot Desk Profile will have the original extension number.)

      When the Change Phone Extension dialog box appears, select the new extension. For details, refer to the DB Programming Help. The phone list is updated to show the converted Hot Desk Profile extension in the Description column and the base phone extension (the new phone extension you just selected) in the Username column, as shown below.

Extension △	Description	Username	Type	A
1015	HD Convert 1014	BASE 1015	86xx	
1016			Digital Telephone	0

    - Click **No** to delete the phone from the list in the Phones folder.

CREATING USERS TO SELECTED PHONES

DB Programming auto-creates Users for the selected phones that are not already used as the main extension for any existing Users. The new Users have the associated phones programmed as the Main Extension, Desk, and Voice Mail destinations. The Last Name and First Name fields are programmed based on the phone's Description field. If any of the selected devices is already programmed as the main extension of a User, a warning message appears.

NOTE	After the User is created, there is no longer a relationship between the Description field and the Last Name and First Name fields. This prevents any special description from being overwritten inadvertently.

See "Description Parsing Rules" on **page 791** for details.

EXPORTING PHONES TO A FILE

Use the Bulk User Export feature to export user information (extension, description, username, and MAC address) from DB Programming into a .csv file to import into Mitel Applications Suite (MAS). The import process parses the description field into First Name and Last Name. The parsing uses the following rules to extract the first and last name:

- If a comma exists in the description, the characters to the left of the comma are extracted as the last name and the characters to the right of the comma are extracted as the first name.

- If there is no comma, but a space is found, the characters to the left of the space are extracted as the first name and the characters to the right of the space are extracted as the last name.

- If there is no comma or space in the description, the entire description becomes the first name and the last name will be blank.

The .csv file contains the following headers:

- ext_phone

- description

- username

- first_name

- last_name

- mac_address

- login

- email_address

- template

The login, email_address, and template fields are left blank in the .csv file because this information is not stored in DB Programming. After you export the information from DB Programming into a .csv file, you can edit the .csv file in a text editor and edit these fields before you import the file into MAS.

The .csv file is not stored in DB Programming. Also, you can use this file to import the user information into DB Programming or MAS. For information about MAS, refer to the *MiVoice Office 250 and NuPoint UM Integration Guide*.

To export user information from DB Programming into a .csv file:

1. Select System – Devices and Feature Codes – Phones – <***phone***>.

2. In the right pane select the phones from which you want to export information into a file.

3. Right-click the selected phones, and then select **Export Phones to File**. The following dialog box appears.

389
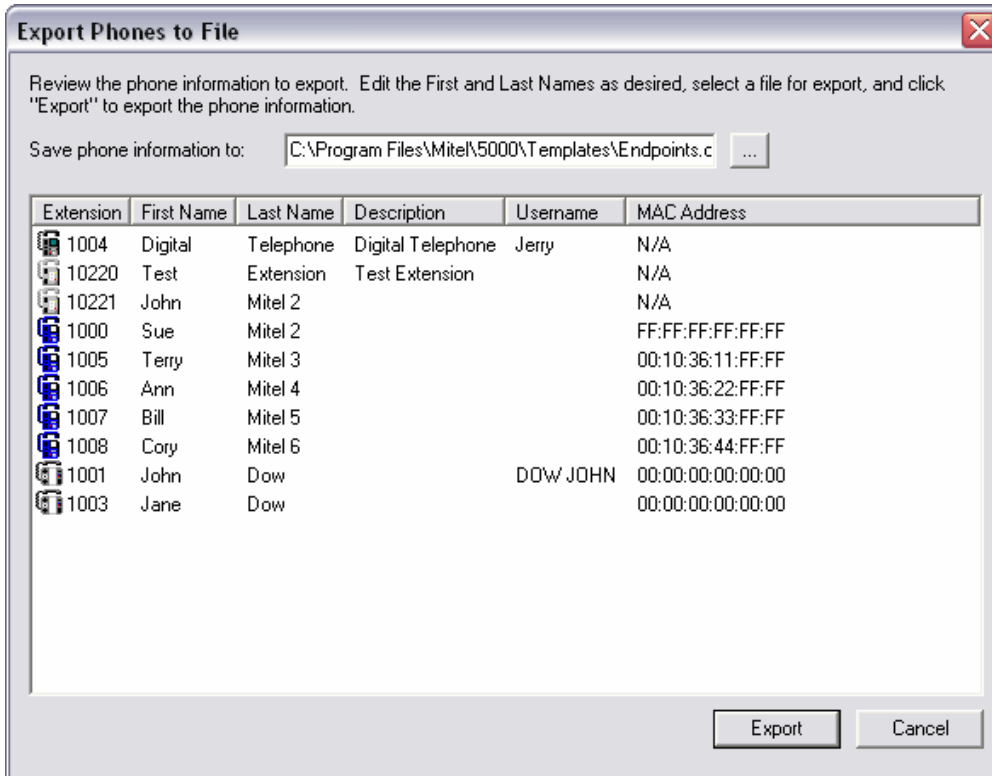
4. Click the First Name column or the Last Name column to edit the name as needed. The name is limited to 255 characters. To edit other information, you need to exit this dialog box and edit the information in DB Programming.

5. In the Save phone information text box, you can leave the default file name (`Phones.csv`), click **Browse** to locate a `.csv` file, or type a file name with a `.csv` extension. The default location where the file is saved is `C:\Users\Public\Documents\MiVoice Office 250` or `C:\Program Files\Mitel\MiVoiceOffice250\Templates` (if the system was updated from v3.0 or earlier)..

6. Click **Export**. A `.csv` file is saved.

7. Import this file into DB Programming or MAS. For information about MAS, refer to the *MiVoice Office 250 and NuPoint UM Integration Guide*.

PROGRAMMING PHONE OPTIONS

You can program the following phone options.

To program phone options:

Select System – Devices and Feature Codes – Phones – Local – *<phone>*.

Table 64:   Phone Programming Option

OPTION \ PHONE TYPE	REF. PAGE	DIGITAL TELEPHONE	SL	52XX/ 53XX	86XX	IP SLA	8602 SOFT-PHONE	SIP PHONE	MICOLLAB CLIENT SOFT-PONE
Associated Extensions	392	✓	✓	✓	✓	✓	✓	✓	✓
Call Logging	395	✓		✓	✓		✓		
Day and Night Classes of Service	400	✓	✓	✓	✓	✓	✓	✓	✓
Phone/Phantom/Hot Desk Profile Flags	401	✓	✓	✓	✓	✓	✓	✓	✓
Forwarding Paths	401	✓	✓	✓	✓	✓	✓		✓
IP Settings	403			✓	✓	✓	✓		✓
Keymaps	403	✓		✓	✓				
Mailboxes	403	✓	✓	✓	✓	✓	✓	✓	✓
Programmable Keys	403	✓		✓	✓				
Record-A-Call	403	✓	✓	✓	✓	✓	✓		✓
Timers (for European systems only)	404		✓						
Account Code Type	404	✓	✓	✓	✓	✓	✓	✓	✓
Validated Flag	404	✓	✓	✓	✓	✓	✓	✓	✓
Languages	406	✓	✓	✓	✓	✓	✓	✓	✓
Secondary Language	407	✓	✓	✓	✓	✓	✓	✓	✓
House Phone and House Phone Day/Night Numbers	408	✓	✓	✓	✓	✓	✓		✓
Passcode	410	✓	✓	✓	✓	✓	✓	✓	✓
Calling Party Name	411	✓	✓	✓	✓	✓	✓	✓	✓
Calling Party Number	411	✓	✓	✓	✓	✓	✓	✓	✓
Emergency Calling Party Number	412	✓	✓	✓	✓	✓	✓	✓	✓
Emergency Dialing Preference	412			✓					

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Table 64:   Phone Programming Option (continued)

OPTION \ PHONE TYPE	REF. PAGE	DIGITAL TELEPHONE	SL	52XX/ 53XX	86XX	IP SLA	8602 SOFT- PHONE	SIP PHONE	MICOLLAB CLIENT SOFT- PONE
Speakerphone Type	**414**	✓							
Attached Device	**414**	✓		✓	✓	✓			✓
Device Audio for Calls Settings	**414**	✓		✓	✓	✓	✓	✓	✓
Echo Profile	**416**	✓		✓	✓	✓	✓	✓	✓
SIP Phone Group	**416**							✓	
Associated Hot Desk Device	**416**	✓		✓	✓		✓		
Application Profile	**416**			✓					

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ASSOCIATED EXTENSIONS

Associated extensions are system extensions used by the phones.

To program Associated Extensions:

1. Select – System – Devices and Feature Codes – Phones – (Local) – *<extension>* – **Associated Extensions**.

2. Configure the following the Associated Extensions:

    *   **Agent Help and Agent Help User-Keyed Extension**: The Agent Help feature allows a phone user to request help from a supervisor during a call. The Agent Help Extension is the number that is called whenever the phone user enters the Agent Help feature code. The Agent Help Extension can be a single phone, or a hunt group, or it can be set to "None." This can be a device located on another node if it is programmed as an off-node device. See "Creating Off-Node Devices" on **page 367**. Optionally, the phone user can be allowed (or required) to enter the Agent Help Extension number, by setting the Agent Help User-Keyed Extension flag, described on **page 393**.

    *   **Alternate Message Source**: If a phone or application has an alternate message source and leaves a message at another phone, the alternate message source is called (instead of the phone or application) when the message recipient responds to the message waiting indications. The alternate message source can be a hunt group, phone, or application. This can be a device located on another node if it is programmed as an off-node device. See "Creating Off-Node Devices" on **page 367**.

    *   **Attendant**: This is the phone, application, or hunt group (if any) that serves as the attendant for this phone. This can be a device located on another node if it is programmed as an off-node device. See "Creating Off-Node Devices" on **page 367**.

    *   **Message Center**: The message center receives messages after the Message Wait timer expires. This can be a device located on another node if it is programmed as an off-node device. See "Creating Off-Node Devices" on **page 367**.

- **Outgoing Extension**: Determines which trunk access (trunk, trunk group, or ARS) this phone uses when an idle CALL button or the OUTGOING button is pressed, or when a System Speed Dial number is selected for dialing before a trunk is selected. Defaults to Use System Outgoing Extension. Phones cannot have direct outgoing access to trunks on other nodes; they must use ARS to access off-node trunks.

- **Transfer Recall Destination**: The transfer recall destination receives transfer calls from this phone. This can be a device located on another node if it is programmed as an off-node device. See "Creating Off-Node Devices" on **page 367**.

- **Voice Mail**: This is the voice mail destination that this phone uses for forwarding calls to voice mail. If this field is not programmed for an Executive or Professional Display telephone or an 8560 digital telephone, the phone will not have the voice mail-related options on the feature button menu display.

  If the Associated Extension Voice Mail field of a phone, phantom device, or Hot Desk profile is edited and the phone, phantom device, or Hot Desk profile is a main extension for a User, the Voice Mail associated destination for the User is automatically updated accordingly. The reverse also applies: if the associated Voice Mail destination of a User is changed, the change will be applied to the Associated Extension Voice Mail field for the main extension phone.
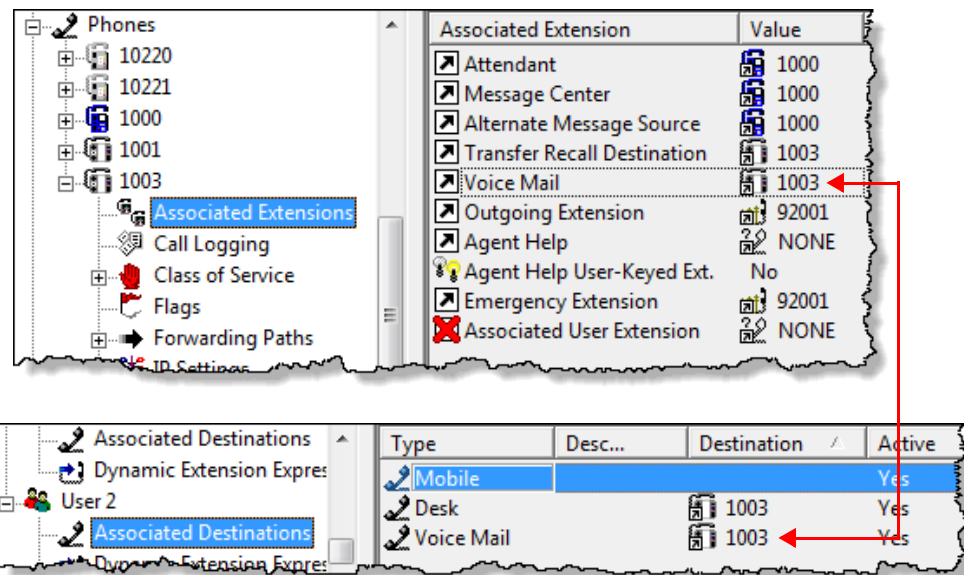
- **Agent Help User-Keyed Extension**: If desired, the phone user can be allowed (or required) to enter the Agent Help Extension number. Enable this flag to allow the phone user to enter the desired extension number. Or, disable the flag to automatically dial the Agent Help Extension programmed (or disable the Agent Help feature for the phone if there is no Agent Help Extension).

a. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

b. Click out of the field or press **ENTER** to save your change. The operation of the feature is determined by the programmed combination of these flags, as shown in **Table 65**.

Table 65:   User-Keyed Extension Examples for Agent Help

AGENT HELP	USER-KEYED EXTENSION	RESULT
Extension number	Disabled	The Agent Help Extension is called when the feature code is entered.
Extension number	Enabled	The user is prompted to enter the desired number after the feature code is entered. If a number is not entered before the dialing timer expires, the programmed Agent Help Extension number is dialed automatically.
None	Enabled	The user is prompted to enter the desired number after the feature code is entered.
None	Disabled	The user hears reorder tone if the feature code is entered.

- **Emergency Extension**: The emergency extension determines which trunk access (trunk, trunk group, or ARS) this phone will use when the Emergency Call phone number is dialed. See the following Notice. This option defaults to Trunk Group 1 (92001). Phones cannot have direct emergency access to trunks on other nodes; they must use ARS to access off-node trunks.

> ## NOTICE
>
> Responsibility for Regulatory Compliance.
>
> It is the responsibility of the organization and persons performing the installation and maintenance of Mitel Advanced Communications Platforms to know and comply with all regulations required for ensuring Emergency Outgoing Access at the location of both the main system and any remote communication phones. Remote IP and SIP phones may require gateway access to nearby emergency responders.
>
> Emergency Call phone numbers include:
>
> - 911, the default for Mitel systems located in the U.S.
> - 999, the default for Mitel systems located in the European market and used primarily in the U.K.
> - If applicable, 112, an emergency number used in Europe outside of the U.K., and Australia.
> - Any emergency number, such as for a police or fire station, that is appropriate for the location of the main system and/or remote phones.

NOTE | If an installation needs Emergency Outgoing Access across nodes, make sure the Local Trunk Group is the first member in the facility group. This allows cross-node emergency calls to use the Local Trunk Group first and not the Remote IP Trunk Group.

- **Associated User Extension**: The Associated User Extension field is a device link with all phone and off-node phone options except for SIP and UC Advanced Softphones. This field is intended to be programmed as the main extension of a User when the specific phone is programmed as one of the Destinations of the User. This field allows a User to use the handoff-push feature from an internal User Destination that is not the main extension.

When the Associated Extensions folder is presented, DB Programming checks to see if there is a User that has the phone as its main extension. If so, the Associated User Extension field for the phone appears with a red "X."

When a phone is associated with a User, then calls to voicemail and other phones will appear to have come from the User's main extension. This allows the end user to call voicemail from one of their non-main extension phones and immediately enter their mailbox. When calling someone from a non-main extension phone, the other person's call logs will show that the main extension called, so that returning the call will call the main extension and therefore use Dynamic Extension Express (DEE) to route the call. When leaving a message from the non-main extension phone on another phone or voicemail, it will appear that the main extension left the message. See "Additional Handoff Push/Pull Destinations" on **page 808** for details.

3. Change the assignment of one of the Associated Phones, using one of the following methods:

Method A

a. Select the current value, then enter the new value in the text box.

b. Press **ENTER**. A screen appears displaying what is associated with the number entered.

c. Click **OK**. The new number appears in the field.

Method B

a. Right-click the existing value. An option box appears.

b. Select **Change**. A window appears prompting for the device type to include.

c. Select the device types (you can use the SHIFT or CTRL key to select more than one item), and then click **Next**. The items with details appear. To view items in a list only, click **List**.

d. Select the device you want to assign as the associated extension, and then click **Finish**. The selection appears in the appropriate field.

NOTE	If the phone and voice mail administrator (refer to the *MiVoice Office 250 Unified Voice Messaging Administrator Guide,* part number 580.8009) adds or changes ring-in devices using the administrator phone, the system automatically changes the ring-in type to Multiple, even when only one device is selected. This occurs because the phone administrator can add multiple devices, which is prevented if the ring-in type is Single.

CALL LOGGING

This section contains the following information:

- **Feature Description** below
- Programming Call Logging on **page 399**

FEATURE DESCRIPTION

The Call Logging feature lists the following types of calls for users of Mitel digital and IP phones and Mitel IP softphone applications:

- Missed Calls
- Received Calls
- Dialed Calls

The Call Logging feature helps you see who called when you were away and makes it easy to redial those people. Non-display and single line phones do not support this feature. Six-line display phones are recommended for field visibility and ease of use.

Benefits of Call Logging include the ability to redial intercom (IC) as well as incoming and outgoing CO calls, to store Caller ID data from incoming calls, and to identify missed calls. The system provides you with an interface similar to cell phones. Call lists are stored in Call Processing and are accessible through the user interface screen on display phones.

For feature usage instructions, refer to the applicable phone user guide.

Call Logging Display

The following graphic shows an example of the Call Logging display on a six-line display phone. Two-line display phones show only the top two lines. Each call entry contains the following fields:

```
JOHN DOE
4809619000
FR AUG 20 11:53A
DELETE   DEL ALL
CALL NOW
<<              >>
```

Name: Displays the name of the missed/received/dialed call. If not available, UNAVAILABLE is displayed.

Number: Displays the number of the missed/received/ dialed call. If not available, UNAVAILABLE is displayed.

Date/Time: Displays the date and time that the call was missed/received/dialed.

DELETE/DELETE ALL: Allows the user to delete the entry indicated on the display or to delete all entries.

CALL NOW: Allows the user to call the number indicated on the display. Note that this feature works on IC calls as well as incoming and outgoing CO calls.

ARROW MENU BUTTONS (<< and **>>):** Appear on a six-line display phone if more than one entry exists on a Call Log list.

A maximum of 20 entries can be stored in each of the three Call Logs associated with a phone. Call Log entries are displayed from newest to oldest. After a Call Log reaches its maximum number of entries, the next call entry appears at the top of the list and the oldest call entry is deleted. Using arrow menu buttons on the six-line display phones or Volume Up and Volume Down buttons, you can scroll through all the entries in a Call Log list. For details, see the following sections.

Call Logging Displays on a Six-Line Display Phone

The following examples show the Call Logging displays that appear on six-line phones.

Idle Menu Display

An option on the Idle Menu of six-line display phones allows you to access the logs directly. Note that the six-line administrator display phones no longer show "ADMIN FEATURES" on

the first line. Rather, it shows only "ADMIN," which makes room for the "MISSED XX" (where XX can be 1–99) calls menu option on the right side of the screen.

<div align="center">

Non-Admin Phone

```
 29999    JOHN DOE
      1:53 FRI AUG 20


OUTGOING      DND
DIRECTORY   LOGS
OTHER FEATURES
```

Admin Phone

```
 29999    JOHN DOE
      1:53 FRI AUG 20

 ADMIN
OUTGOING      DND
DIRECTORY   LOGS
OTHER FEATURES
```

</div>

Main Menu Display

When the LOGS menu option is selected, the following display appears.

<div align="center">

```
LOG TYPE: MISS = 1
RCV=2 DL=3 CLR=4

MISSED CALLS
RECEIVED CALLS
DIALED CALLS
CLEAR CALL LOGS
```

</div>

Call Logging Displays

On a six-line display phone, only the top two lines show Call Logging information. When you select option 1, 2, or 3 from the main menu, displays similar to the following examples appear.

<div align="center">

**Calling number
and name**

```
JOHN DOE
4809619000

FR AUG 20 11:53A

DELETE   DEL ALL
CALL NOW
<<            >>
```

**Calling number
only**

```

4809619000

FR AUG 20 11:53A

DELETE   DEL ALL
CALL NOW
<<            >>
```

**No calling number
or name**

```
UNKNOWN CALLER


FR AUG 20 11:53A

DELETE   DEL ALL
CALL NOW
<<            >>
```

</div>

Missed Calls

The Missed Calls feature is supported only on six-line display phones. If a phone has registered missed calls that the user has not yet viewed, the MISSED <1–99> calls menu option appears on the display, as shown in the following example.

<div align="center">

Non-Admin Phone   **Admin Phone**

29999   JOHN DOE
1:53 FRI AUG 20
MISSED 10
OUTGOING   DND
DIRECTORY LOGS
OTHER FEATURES

29999   JOHN DOE
1:53 FRI AUG 20
ADMIN **MISSED 10**
OUTGOING   DND
DIRECTORY LOGS
OTHER FEATURES

</div>

When the user selects the MISSED XX calls menu option from an idle display, the user is taken directly to the MISSED Calls Menu, bypassing the LOGS menu.

NOTE	If using a six-line display phone in conjunction with the Unified Communicator (UC) application in a network environment, missed calls may be registered accurately only in the UC Call Log. Missed calls that have been routed across nodes may show as Received rather than Missed. In this situation, the user should rely on the UC Call Log for an accurate view of missed calls.

Once the Missed Calls menu option is selected, the Idle Display menu no longer shows MISSED <1–99> on the display, regardless of whether the user actually looks at every one of the missed calls.

To access the Missed Calls log at any time, the user can either press the LOGS menu button next to the phone display screen or press the Special button and enter the Call Logging feature code, 333. The screen displays the LOGS menu, and from there the user can navigate to the Missed Calls menu.

Call Logging Displays on a Two-Line Display Phone

The following examples show Call Logging displays that appear on a two-line display phone.

NOTE	At any menu level, you can press the asterisk (*) button to cancel or return to the previous menu, or press the pound (#) button to accept.

Idle Menu Display

The following example shows a two-line display when the phone is idle.

<div align="center">

21533
JOHN SMITH
WE OCT 23 09:17A

</div>

Main Menu Display

To access the Call Logging feature, press the Special button and enter 333, the Call Logging feature code. The following display screen prompts you to enter 1 for Missed (MISS) calls, 2 for Received (RCV) calls, 3 for Dialed (DL) calls, or 4 to clear (CLR) all call logs.

<div align="center">

LOG TYPE:
MISS=1
RCV=2 DL=3 CLR=4

</div>

Call Logging Displays

On a two-line display phone, only the top line shows Call Logging information.When you select option 1, 2, or 3 from the main menu, displays similar to the following examples appear.

Calling number and name	**Calling number only**	**No calling number or name**
4809619000 JOHN WE OCT 23 09:17A	4809619000 WE OCT 23 09:17A	UNKNOWN CALLER WE OCT 23 09:17A

PROGRAMMING CALL LOGGING

Table 66 shows Call Logging options. Single line phones do ***not*** support Call Logging.

Table 66:   Call Logging Options and Descriptions

FIELD NAME	RANGE	DEFAULT	DESCRIPTION
Allow Duplicate Entries	Yes/No	Yes	Determines whether or not duplicate entries may appear in the call logs.
Enable Call Logging			Determines whether or not Call Logging will be enabled for a particular phone.
Maximum Number of Entries	0–20	10	Describes the maximum number of entries that will be stored in each log.

When the **Enable Call Logging** field is set to "No," the other two Call Logging fields display a red "X." This indicates all features associated with Call Logging are also disabled. The following example shows the Call Logging fields for a digital telephone.
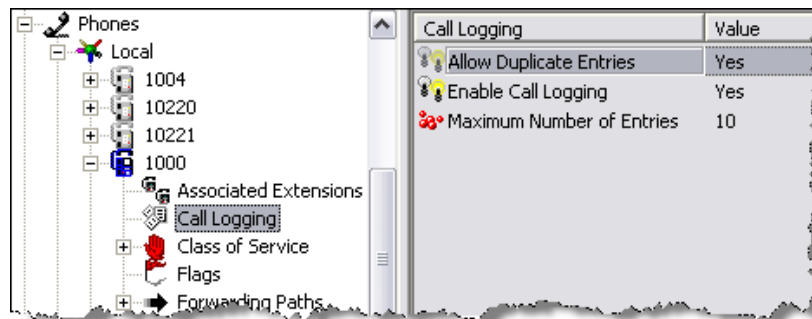
The copy/paste functionality for phones includes a Call Logging option. This functionality will copy/paste the field values for the three Call Logging fields described in **Table 66**. Call Logging is not a copy/paste attribute for single line phones.

By default, a 1 is added to the beginning of returned CO calls. If needed, be sure to allow local Area Codes in User Group 1.

To change Call Logging options:

1.   Select – System – Devices and Feature Codes – Phones – (**Local**).

2. Select the extension number.

3. Select **Call Logging**.

4. Enable the options as described in **Table 66.**

5. Click out of the field or press **ENTER** to save your changes.

DAY AND NIGHT CLASSES OF SERVICE

Each phone has Classes of Service (COS) which restrict or allow certain dialing patterns from being dialed on a call. For details, see **page 531** for COS and **page 527** for toll restriction.

PROGRAMMING PHONE TOLL RESTRICTIONS

To program phone toll restrictions:

1. Select – System – Devices and Feature Codes – Phones – (**Local**).

2. Select the extension number.

3. Select **Day** (or **Night**) **Class of Service**.

4. Double-click the desired time period to view a list of current classes of service.

To add a class of service:

a. Right-click in the window, and then select **Add To List**.

b. A window appears that allows you to select toll restriction types. Select **ARS Only & Deny Area/Office** and/or **Classes of service**, and then click **Next**.

c. The Classes of Service appear.

d. Select the desired classes of service, then select **Add Items**. The selected classes of service appear in the list. Click **Finish** to exit.

e. *(U.S. only) If you selected Deny Area/Office class of service,* the phone must also be assigned to a User Group. To change the user group, right-click **User Group**, then select **Change User Group**. In the first window that appears, select **User Groups**, then click **Next**. In the next window, select the desired user group, and then click **Finish** to exit and save the change.

DELETING CLASSES OF SERVICE

To delete one or more class of service:

1. Select – System – Devices and Feature Codes – Phones – (**Local**).

2. Select the extension number.

3. Select **Day** (or **Night**) **Class of Service**.

4. Right-click, and then select **Remove Selected Items**. To select a series of items, hold down **SHIFT** while selecting the first and last item in the range. To select two or more that are not consecutive, hold down **CTRL** while selecting the desired items.

PHONE/PHANTOM/HOT DESK PROFILE FLAGS

See **page 417** for details.

FORWARDING PATHS

Each phone can have up to three forwarding paths, and there can be 200 different programmed paths in the system, numbered 001–200. Path 000 (No Forwarding Path) can be assigned to disable system forwarding for the phone. For more details on system forwarding, see "Forwarding – System Forwarding" on **page 298**.

The following are Forwarding Path types:

- **Forwarding Path**: Shown for reference only. To view the programming for the path, double-click Forwarding Paths to show Phone-Related programming.

- **Ring Principal Once**: If the system-forwarded device is a phone and the user wants to be notified each time a call is sent to the system forwarding path, enable this option. The phone user hears a single ring tone each time a call is system forwarded and the display phone shows CALL SENT TO FORWARD PATH. In the default state, this is disabled.

- **Phone Conditions (Busy, DND, No Answer, and Immediate)**: The phone can be programmed to use the path to forward calls when the phone condition is busy, in Do-Not-Disturb, and/or no answer. Or, calls can be forwarded immediately after they are received at the phone. In the default state, all options but Immediate are selected. Determine which phone conditions will cause the calls to follow the forwarding path and enable them. Note that if any condition other than Immediate is selected, Immediate will appear dimmed and cannot be selected.

- **System Conditions (Day and Night Mode)**: You can program the phone to use the forwarding path when the system is in day mode and/or night mode. By default, both day and night mode are selected. Indicate whether calls follow the forwarding path in day and/or night mode by enabling the desired system conditions.

- **Forwarding Call Types**: The types of calls that can be forwarded are listed below. The forwarding phone can be programmed to forward any or all of these call types. (In the default state, call routing, ring-in, DISA, and transferred calls are enabled.) Determine the types of calls that will follow the forwarding path by enabling each desired call type. Available options include:

  - *Call Routing Table*: Outside calls received through a call routing table, including DID and E&M [DDI] calls, but not including DISA calls.

  - *CO Ring-In*: Ringing outside calls, including calls received through network connections.

  - *CO Transfer/AA/VM*: Transferred outside calls, including automated attendant and voice mail transfers and transfers from other nodes.

  - *CO Recall*: Recalling outside calls, including calls from other nodes.

  - *DISA*: DISA calls, including DISA calls received through a call routing table.

  - *IC Calls*: Intercom calls, including calls from other nodes.

ADDING FORWARDING PATHS

To add a forwarding path:

1. Select – System – Devices and Feature Codes – Phones – (**Local**).

2. Select the extension number.

3. Select **Forwarding Paths**.

4. Right-click anywhere in the right side of the window, and then click **Add To Forwarding Paths List**. A window appears prompting for the device types to include.

5. Select the forwarding paths (you can use the SHIFT or CTRL key to select more than one item), and then click **Next**. The items with details appear. To view items in a list only, click **List**.

6. Select the appropriate paths, then select **Add Items**. When you have added all the paths you want, click **Finish**. The selections appear in the list.

DELETING FORWARDING PATHS

To delete a path:

1. Select – System – Devices and Feature Codes – Phones – (**Local**).

2. Select the extension number.

3. Select **Forwarding Paths**.

4. Select the path, right-click, and then select **Remove Selected Items**.

PROGRAMMING SPECIFIC FORWARDING PATHS

NOTE	Call forwarding paths are checked in numerical order when they apply to the same call type (for example, multiple forwarding paths for outside calls). If one forwarding path is for the "No Answer" call condition, it should be last on the list of phone forwarding paths. If No Answer is the condition in the first path, the system always waits for the System Forward Initiate timer to expire before forwarding, even if the phone is in Do-Not-Disturb mode or busy.

To program a specific Forwarding Path:

1. Select – System – Devices and Feature Codes – Phones – (**Local**).

2. Select the extension number.

3. Select **Forwarding Paths**.

4. Double-click the forwarding path to view the following options:

ENABLING FORWARDING PATH OPTIONS

To enable a Forwarding Path option:

1. Select – System – Devices and Feature Codes – Phones – (**Local**).

2. Select the extension number.

3. Select **Forwarding Paths**.

4. Select the option.

5. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

6. Click out of the field or press **ENTER** to save your change.

IP SETTINGS

See "Phone and Device IP Settings" on **page 641** for details.

KEYMAPS

See **page 425** for details.

MAILBOXES

If the phone being programmed serves as the message notification phone for one or more mailboxes, the mailboxes appear in the Mailboxes list.

To view the Mailboxes list:

1. Select – System – Devices and Feature Codes – Phones – (**Local**).

2. Select the extension number.

3. Select **Mailboxes** to see the list of mailboxes assigned to this phone. You cannot program the mailbox here.

4. Double-click the mailbox. The screen jumps to the Mailbox programming area.

PROGRAMMABLE KEYS

See **page 439** for details.

RECORD-A-CALL

You must create the Record-A-Call application the Voice Processor database before you can enable the feature. See "Record-A-Call" *on* **page 893***.*

There are three fields for the Record-A-Call feature: Mailbox, Mailbox User-Keyed Extension Flag, and Application. The application and mailbox might not be on the same node as the phone. If so, you must program the voice processor applications as off-node devices on the local node, and the phone must be an off-node device on the Voice Processor node. See "Creating Off-Node Devices" on **page 367**.

Determine which mailbox, if any, is dialed automatically when the Record-A-Call feature is used. The Record-A-Call Mailbox can be set to "This Phone's Associated Mailbox" to call the mailbox assigned to that phone, or it can be set to any valid mailbox number. If you do not want a mailbox number dialed automatically when the Record-A-Call feature is used at this phone, enable the User-Keyed Mailbox flag. This overrides the automatic entry and allows the phone user to enter the desired mailbox number. The operation of the feature is determined by the programmed combination of these flags, as shown in **Table 67**.

Table 67:   Record-A-Call Operation

RECORD-A-CALL MAILBOX	USER-KEYED MAILBOX	RESULT
"Associated" or mailbox number	Disabled	The Record-A-Call Mailbox is automatically called when the feature code is entered.
"Associated" or mailbox number	Enabled	The user is prompted to enter the desired mailbox number after the Record-A-Call feature code is entered. If a number is not entered before the dialing timer expires, the programmed Record-A-Call Mail box number is dialed automatically.

TIMERS

(*These timers are applicable to single line phones in European systems only.*) In the U.S., this folder is only available in Online Monitor mode.

The system supports the transmission of Caller ID and Calling Line Identification (CLID in Europe) to single-line sets in Europe. This feature uses the calling party information that the system unit receives from the local network provider. After programmed, on-hook single-line sets display the calling party's information when receiving an incoming outside call. The Caller ID (CLID) information is also displayed if the single-line set receives a transferred call from another phone that has calling party information.

To use Caller ID (CLID) for single-line sets:

1. Enable the Caller ID flag (System\Devices and Feature Codes\Phones\Local\<*Single-Line Ext*>\Flags).

2. Program the following timers:

    •   **Caller ID Line Reversal Time**: Enter the number of milliseconds that the system reverses the line polarity so that the single line set can receive Caller ID (CLID) information. This value may differ for various phones. The valid range is 0–200 milliseconds; the default value is 30 milliseconds.

    •   **Caller ID Alerting Timer**: Enter the number of milliseconds that the system will send Caller ID (CLID in Europe) alert tones to the single line set. These tones are not audible to the end user and are used strictly to notify the phone that information will be transmitted. This value may differ for various phones. The valid range is 88–110 milliseconds; the default value is 100 milliseconds.

ACCOUNT CODE TYPE

See **page 453** for details.

VALIDATED FLAG

See **page 456** for details.

PROGRAMMING THE RECORD-A-CALL MAILBOX

You can assign the mailbox to which Record-A-Call mailbox records and saves calls.

To program the Record-A-Call mailbox:

1. Select – System – Devices and Feature Codes – Phones – (**Local**).

2. Select the extension number.

3. Select **Record-A-Call**.

4. Use one of the following methods:

    Method A

    a. Select the current value, then enter the new value in the text box.

    b. Press **ENTER**. A screen appears displaying what is associated with the number entered.

    c. Click **OK**. The new number appears in the field.

    Method B

    a. Right-click the current Value. An option box appears.

    b. Select **Change Mailbox**. A window appears prompting for the mailbox type to include.

    c. Select **Mailbox**, **This Phone's Associated Mailbox**, **Off-Node Mailbox,** or **Unassociated Mailbox Off-Node Device**, then click **Next**. The list of mailboxes appears. You can view them in a list by selecting the List button or view details by selecting the Details button.

    d. Select the desired mailbox, then click **Finish**. The selection appears in the Mailbox field.

PROGRAMMING THE MAILBOX USER-KEYED EXTENSION

The Mailbox User-Keyed Extension option determines the mailbox destination for recorded calls. If the value is "No" (disabled), the system uses the mailbox stored in the Record-A-Call "Mailbox" field (see the previous section). If enabled, the system prompts the user for a mailbox number when Record-A-Call is activated. The default value is No (disabled).

To set the Mailbox User-Keyed Extension option:

1. Select – System – Devices and Feature Codes – Phones – (**Local**).

2. Select the extension number.

3. Select **Mailbox User-Keyed Extension**.

4. In the Value column, select the check box. The field changes to **Yes**. To disable the flag, clear the check box.

5. Click out of the field or press **ENTER** to save your change.

PROGRAMMING THE RECORD-A-CALL APPLICATION

The voice processor must have one or more applications created for the Record-A-Call feature. You can choose the application that is used by the phone. If you choose **None**, the phone will not have access to the Record-A-Call feature.

To set the Application:

1. Select – System – Devices and Feature Codes – Phones – (**Local**).

2. Select the extension number.

3. Select **Application**.

4. Use one of the following methods:

   Method A

   a. In the Value column, select the current value, then enter the new value in the box.

   b. Press **ENTER**. A screen appears displaying information associated with the number entered.

   c. Click **OK**. The new number appears in the field.

   Method B

   a. Right-click the current value. An option box appears.

   b. Select **Change Application**. A window appears prompting for the application type to include.

   c. Select **None**, **Record-A-Call**, or **Off-Node Record-A-Call**, then click **Next**. The list of applications with details appear. To view the applications in a list only, click **List**.

   d. Select the desired application, and then click **Finish**. The selection appears in the Application field.

LANGUAGES

You can set the (primary) languages that display for the voice prompts and phone displays. This field can be set to any specific language so that the system can support more than two languages. The language choices are Use Primary Language, Use Secondary Language, American English, British English, Canadian French, Japanese, or Spanish. End users can also select the primary or secondary language, if enabled. See "Secondary Language" on **page 407**.

> **NOTE** 52xx\53xx phones do not support the Japanese Language.

For complete information about multilingual capability, see "Multilingual Capability" on **page 314**.

To select a primary language:

1. Select – System – Devices and Feature Codes – Phones – (**Local**).

2. Select the extension number.

3. Select **Language**.

4. In the **Value** column, select the option from the list.

5. Click out of the field or press **ENTER** to save your change.

SECONDARY LANGUAGE

The Secondary Language option corresponds to the Change Language feature (301). The Change Language feature is used to toggle between the system Primary Language and the system Secondary language.

> **NOTE** 52xx\53xx phones do not support the Japanese Language.

It toggles between the System Primary Language and the phone Secondary Language, or can specify a language. This allows any phone in the system to have its own secondary language or use the System Secondary Language, giving the system the ability to support more than two languages.

- If the phone Secondary Language field is programmed to be Use Primary Language the Change Language feature will do nothing because the phone will toggle between the System Primary Language and the phone Secondary Language, which is the System Primary Language.

- If the phone Secondary Language field is programmed to be Use Secondary Language the Change Language feature will toggle between the System Primary Language and the phone Secondary Language, which is the System Secondary Language. This is the system default.

- If the phone Secondary Language field is programmed to be Japanese the Change Language feature toggles between the System Primary Language and the phone Secondary Language, which is Japanese.

The Language field for phones indicates what language the phone is currently set to. This field used to toggle between the Use Primary Language and Use Secondary Language. It can be set to any specific language along with the Use Primary Language and Use Secondary Language.

The side-effect of changing the Language field is that if you change it to Japanese and the phone Secondary Language field is set to Spanish, then the user will have no way to get back to Japanese if they enter the Change Language feature code. This is because the first time the user enters the feature code, the system toggles the phone to the Use Primary Language option. The next time the user enters the feature code, the system toggles the language to the phone Secondary Language, which is Spanish. To avoid this situation, simply change the phone Secondary Language to Japanese and the phone Language field to Japanese.

For complete information about multilingual capability, see "Multilingual Capability" on **page 314**.

To select a secondary language:

1. Select – System – Devices and Feature Codes – Phones – (**Local**).

2. Select the extension number.

3. Select **Secondary Language**.

4. In the **Value** column, select the option from the list.

5. Click out of the field or press **ENTER** to save your change.

HOUSE PHONE

This section contains the following information:

- **Feature Description** below

- Programming House Phones on **page 409**

FEATURE DESCRIPTION

This feature provides users with the ability to place a predesignated intercom or outside call simply by lifting the handset (or pressing the Speaker button, if using a phone) on a designated House Phone. In a network, the House Phone can be programmed to dial an off-node device. There are several applications for this feature, such as:

- **Courtesy paging phone:** Visitors hear pages instructing them to pick up the House Phone (such as the paging phones used in airport terminals). When they lift the handset, they are connected to a pre-programmed phone user who can give them a message or connect them to a call.

- **Emergency phone:** The House Phone can be programmed to automatically dial the Emergency Call feature code. This could save time in an emergency. For details about the feature, see "Emergency Calls" on **page 182**.

- **Service phone:** Customers can use the House Phone(s) to place orders or receive special services from the lobby. For example, the House Phone would automatically dial the extension number of a service representative (or hunt group number of the service department).

- **Intercom network:** House Phones could be placed in strategic locations throughout a building—for example, in hazardous areas—and programmed to call a specific phone or group of phones, such as environmental safety or security offices.

The primary attendant should not be a House Phone.

The number dialed by the House Phone is determined by the phone's Speed Dial programming. The number programmed through the database or in Station Speed Dial location 0 is automatically dialed during day mode, and the number in location 1 is dialed during night mode. This number can be either an extension number or an outside telephone number. If it is an outside number, it must be preceded with a trunk access code (and a pause if necessary).

After the House Phone status has been programmed, the Speed Dial number can be changed only while on-hook (if it is a phone) or through individual phone information (special purpose phone) programming because lifting the handset will cause the phone to dial the designated number.

Incoming calls take precedence over outgoing calls. If using a single line phone or a phone that is programmed for automatic trunk access, any ringing call is automatically answered when the handset is lifted or the Speaker button is pressed.

The "House Phone Mode" flag determines whether a single line House Phone returns dial tone or does not return dial tone after the called party disconnects. At the System level of programming, all House Phones may be set in either Normal mode or Restricted mode. At the individual Phone level of programming, a House Phone must be programmed to dial specific

digits as soon as the handset is taken off-hook. The programmed digits may ring a specific phone or a Hunt Group that rings multiple phones.

Normal mode allows the user to enter a feature code or place a call after the automatically called number hangs up. Restricted mode prevents the user from performing any operation other than placing a House Phone call. The System default state is Normal.

The interaction of the House Phone Mode flag with the programmed Speed Dial number and system feature is shown in **Table 68**.

Table 68:   House Phone Day/Night Functions and Normal/Restricted Modes

SITUATION	DAY/NIGHT #	NORMAL MODE	RESTRICTED MODE
House Phone user lifts the handset	Complete extension	Dials the extension number	Dials the extension number
House Phone user lifts the handset	Blank	User receives intercom dial tone	User receives reorder tone
House Phone user lifts the handset	Incomplete extension number	System dials the partial number and waits for further digits	System dials partial number, then times out after Long Interdigit timer expires, and sends reorder tones
House Phone is connected to ARS and the user performs a hookflash	Dials an outside number using ARS	System inserts a hookflash in the number and registers end-of-dialing	System restarts ARS by clearing the number and returns outside dial tone to the user
House Phone is connected to an outside call and the user performs a hookflash	N/A	The call is placed on consultation hold and the user hears intercom dial tone	The trunk dials a hookflash and toll restriction is restarted
Party the House Phone is connected to hangs up before the House Phone user	N/A	User receives intercom dial tone and can use another feature code or dial a number	System re-dials the House Phone digits after the SL Wait for Disconnect timer expires

PROGRAMMING HOUSE PHONES

This section describes how to program house phones.

Assigning a Phone as a House Phone

You can assign a phone as a house phone.

To enable the House Phone flag and assign a phone as a House Phone:

1. Select – System – Devices and Feature Codes – Phones – (**Local**).

2. Select the extension number.

3. Select **House Phone**.

4. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

5. Click out of the field or press **ENTER** to save your change.

Assigning House Phone Day and Night Extension Numbers

The assigned day number is dialed when the system is in day mode and the assigned night number is dialed when the system is in night mode. You can also program the day or night House Phone numbers using the House Phone Speed Dial locations.

To program House Phone day and night numbers:

1. Select – System – Devices and Feature Codes – Phones – *<extension>* – **House Phone Day** (or **Night**).

2. In the **Value** column, enter the extension or outside number (up to 16 digits, including pauses and hookflashes [recalls] that are dialed when the House Phone is used during the day and/or night mode).

3. Click out of the field or press **ENTER** to save your change. You can use the extension number of an off-node device, if desired. These numbers are also phone speed-dial locations codes 0 (day) and 1 (night).

PASSCODE

You can change the passcode for Remote Programming or SIP peer SIP phones. The default passcode is the extension number of the phone. The passcode can also be changed by entering the Program Passcode feature code at the phone or when using the Remote Programming feature.

For more information about the Remote Programming feature, see "Remote Programming" on **page 335**. For more information about SIP phones, see "SIP Phones and SIP Phone Groups" on **page 673**.

To program a password for the phone:

1. Select – System – Devices and Feature Codes – Phones – (**Local**).

2. Select the extension number.

3. Double-click **Passcode**. The Edit Passcode dialog box appears.

4. In the **New Passcode** box, type the new passcode (up to 8 digits). Typed characters appear as asterisks (***). For SIP phones, the SIP phone's passcode must match the User Passcode.

5. Retype the passcode exactly as before in the **Confirm Passcode** box.

6. Click **OK** to exit and save the passcode. If the entered passcodes match, you will return to the Passcode field. If not, you must re-enter the new passcode and verify it again. If you make a mistake while entering the passcode or want to leave it unchanged, select **Cancel**. To prevent unauthorized use of call forward to the public network, all phones using Remote Programming should have a passcode. To make the passcodes difficult to guess, they should not match the extension number or consist of one digit repeated several times.

CALLING PARTY NAME

This field is similar to the existing Calling Party Number field. It is used only for ISDN or SIP peer trunk calls to the public network (non-private networking). If this field is set, the system may use this information for the outgoing ISDN or SIP setup request message. You can program up to 16 alphanumeric characters (up to 15 alphanumeric characters, except for :, ", and " for a voice mail or SIP voice mail application) in the box. This field is not used when the "Send Station Extension/Username to Attached PBX" on **page 522**, "Propagate Original Caller ID" on **page 523**, and/or "Propagate Original Caller ID on Transfer" on **page 919** (for voice mail) flags are enabled.

To enter a Calling Party name:

1. Do one of the following:

    -   System – Devices and Feature Codes – Phones – Local – <***phone***>.

    -   System – Devices and Feature Codes – SIP Peers – SIP Trunk Groups – <*SIP trunk group*> – **Trunk Group Configuration**.

    -   System – Devices and Feature Codes – SIP Peers – SIP Voice Mails – <SIP voice mail> – Applications – <***application***>.

    -   Voice Processor – Devices – Applications – <***application***>.

2. Select **Calling Party Name***.*

3. In the **Value** column, select the current value, and then type the name in the box.

4. Click out of the field or press **ENTER** to save your change.

CALLING PARTY NUMBER

Depending on how flags are set, each phone, CO trunk group and/or SIP peer trunk group can be programmed to send an identifying number when a call is placed. This is called the "Calling Party Number." (This information is required for emergency calls in some states.) You can program any number up to 48 numeric characters (up to 15 alphanumeric characters, except for :, ", and " for a voice mail or SIP voice mail application) in the Calling Party Number field. However, check with your service provider to determine their specific requirements for this field. This number will be sent in the ISDN or SIP setup message in the Calling Party Number Information Element. In addition, the system will also send the extension number of the phone in the Calling Party Number Subaddress Information Element. The CO (local exchange) should ignore this information element if it does not support it. See **page 534** for details about Caller ID Forwarding. In the case of SIP, it will influence the "From," "Contact," and "P-Asserted-Identity" fields in the SIP headers.

NOTE	There is no default number for this field. It is up to you to supply the correct Emergency Calling Party Number for each phone.

To enter a number to display as the Calling Party Number:

1. Do one of the following:

    -   System – Devices and Feature Codes – Phones – Local – <***phone***>.

    -   System – Devices and Feature Codes – SIP Peers – SIP Trunk Groups – <*SIP trunk group*> – **Trunk Group Configuration**.

- System – Devices and Feature Codes – SIP Peers – SIP Voice Mails – *<SIP voice mail>* – Applications – *<**application**>*.

- Voice Processor – Devices – Applications – *<**application**>*.

2. Select **Calling Party Number***.*

3. In the **Value** column, select the current value, and then type the number in the box.

4. Click out of the field or press **ENTER** to save your change.

EMERGENCY CALLING PARTY NUMBER

You can specify the Caller ID [CLID] to use when a phone makes an outgoing emergency call. Call Processing propagates the appropriate calling party number to the PSTN. This field is similar to the existing "Calling Party Number" field. It is used only for ISDN calls to the public network (non-private networking). If this field is programmed, the system may use this information for the outgoing ISDN setup request message. You can program up to 20 alphanumeric characters in the box. This field is not used when the "Send Station Extension/ Username to Attached PBX" on **page 522** or "Propagate Original Caller ID" on **page 523** flags are enabled. See **page 534** for details about Caller ID Forwarding.

To enter an Emergency Calling Party Number:

1. Select – System – Devices and Feature Codes – Phones – (**Local**).

2. Select the extension number.

3. Select Emergency **Calling Party Number***.*

4. In the **Value** column, select the current value, and then type the number in the box.

5. Click out of the field or press **ENTER** to save your change.

EMERGENCY DIALING PREFERENCE

The Line Interface Module (LIM) is an optional device that adds analog trunk capability to the Mitel 5224, 5324, 5330, 5340, and 5360 IP phones. In the event of a network failure, the phone can be configured to use the public switched telephone network (PSTN) to provide continued basic telephony until service in restored. The analog line can also be used for emergency calls and local calling.

The LIM can be used to:

- allow incoming and outgoing analog PSTN calls directly to and from the IP phone

- provide survivability (failover) in the event IP connectivity is lost

- provide emergency dialing support (emergency calls can be programmed to connect through the analog PSTN connection)

NOTE	When using the analog line attached to the LIM, the phone's functionality is reduced to that of a basic telephone without any features. For example, users are unable to change volume, mute calls, transfer calls, put calls on hold, etc. Also, keep the following considerations in mind when using the LIM: • If the LIM line rings in while there are multiple calls on hold, the first call placed on hold is disconnected. • When the LIM line rings in, the phone beeps once every so often to indicate the incoming call. • No display information is available for the LIM analog line. For additional considerations, refer to the <u>Mitel Knowledge Base Center</u> (www.inter-tel.com/knowledgebasecenter).

Each IP phone equipped with a LIM can be programmed with one of the following emergency call routing preferences:

• **System Only:** Emergency calls are routed using the system's designated emergency call routing rather than using the analog trunk attached to the LIM. This is the default emergency call handling rule for all phones that do not have a LIM attached.

• **LIM First:** Emergency calls are first routed through the LIM. If unsuccessful, calls will then be routed through the system.

• **LIM Only:** Emergency calls are routed through the LIM only. If unsuccessful, the display indicates that the call failed.

NOTE	When using the analog line attached to the LIM, the phone's functionality is reduced to that of a basic telephone without any features. For example, users are unable to change volume, mute calls, transfer calls, put calls on hold, and so forth.

To program the LIM:

1. Select System – Devices and Feature Codes – Phones – *<52xx/53xx IP phone>* – **Emergency Dialing Preference**.

2. Select one of the following type from the drop-down list:

   • Line Interface Module First

   • Line Interface Module Only

   • System Only (default)

The field cannot be copied/pasted between phones because it is specific to the phone as to whether or not a LIM can be attached or is attached.

If the "Line Interface Module Only" option is selected, emergency calls fail if a LIM is not attached to the phone or if the phone is a model type that does not support the LIM. Whenever this option is selected, DB Programming displays a warning indicating that emergency calls will only go through a LIM.

SPEAKERPHONE TYPE

Speakerphone Type

(**Not** *used on single line phones.*) This flag affects digital telephones differently:

- **Executive or Professional Display Phone**: To enable the integrated speakerphone, to be used in standard and/or enhanced mode, select either of the speakerphone options (Enhanced or Standard). Selecting Standard does not prevent the Executive or Professional Display user from using the enhanced speakerphone. To disable the Executive or Professional Display speakerphone, select **None**.

- **Other Digital Telephones**: If the telephone will have access to shared standard speakerphone resources, select **Standard**. For access to enhanced and standard speakerphone resources, select **Enhanced**. To prevent any of these telephones from using shared speakerphone resources, select **Disabled**.

ATTACHED DEVICE

(**Not** used for single line phones.) Indicate the type of attached device connected to the phone. Or, if nothing is connected, select **None**. The other option is **Desktop Interface Socket**.

To set the attached device field:

1. Select – System – Devices and Feature Codes – Phones – (**Local**).

2. Select the extension number.

3. Select **Attached Devices**.

4. In the **Value** column, select the option from the list.

5. Click out of the field or press **ENTER** to save your change.

DEVICE AUDIO FOR CALLS SETTINGS

You can program the following audio options that callers hear when waiting for system users:

- **Audio for Calls Camped onto this Device** below
- **Audio for Calls Holding for this Device** below
- Audio for Calls Ringing this Device on **page 415**

AUDIO FOR CALLS CAMPED ONTO THIS DEVICE

You can select the audio that callers hear when camped-on to the phone.

To program the Audio for Calls Camped onto this Device:

1. Select – System – Devices and Feature Codes – Phones – (**Local**).

2. Select the extension number.

3. Select **Audio for Calls Camped onto this Device**.

4. Select one of the following options from the list:

   - **Silence**: Callers hear no Music-On-Hold.

   - **Tick Tone**: Callers hear tick tone.

   - **Ringback**: Callers hear ringback.

   - **MOH Port**: Callers hear an external music source. This is the default value.

   - **File-Based MOH**: Callers hear the MOH file selected in DB Programming. For more information, see "File-Based Music-On-Hold (MOH)" on **page 741**.

5. Click out of the field or press **ENTER** to save your change.

AUDIO FOR CALLS HOLDING FOR THIS DEVICE

You can select the audio that callers hear when placed on hold at the phone.

To program Audio for Calls Holding for this Device:

1. Select – System – Devices and Feature Codes – Phones – (**Local**).

2. Select the extension number.

3. Select **Audio for Calls Holding for this device**.

4. Select one of the following options from the list:

   - **Silence**: Callers hear no Music-On-Hold.

   - **Tick Tone**: Callers hear tick tone.

   - **Ringback**: Callers hear ringback.

   - **MOH Port**: Callers hear an external music source. This is the default value.

   - **File-Based MOH**: Callers hear the MOH file selected in DB Programming. For more information, see "File-Based Music-On-Hold (MOH)" on **page 741**.

5. Click out of the field or press **ENTER** to save your change.

AUDIO FOR CALLS RINGING THIS DEVICE

You can select the audio that callers hear when ringing the phone.

NOTE	The Audio for Calls Ringing this Device option only works when the call goes through a trunk group and also when used in conjunction with the Use Next Device's Audio Source field. IC calls do not apply to the use of this field when this field is set to a music source. For a hunt group in which the primary purpose is to support IC callers (for example, an internal help desk), you should set all of the "Audio for Calls..." fields to something other than a music source, such as Ringback.

To program Audio for Calls Ringing this Device:

1. Select – System – Devices and Feature Codes – Phones – (**Local**).

2. Select the extension number.

3. Select **Audio for Calls Ringing this device**.

4. Select one of the following options from the list:

   - **Silence**: Callers hear no Music-On-Hold.

   - **Tick Tone**: Callers hear tick tone.

   - **Ringback**: Callers hear ringback. This is the default value.

   - **MOH Port**: Callers hear an external music source.

   - **File-Based MOH**: Callers hear the MOH file selected in DB Programming. For more information, see "File-Based Music-On-Hold (MOH)" on **page 741**.

5. Click out of the field or press **ENTER** to save your change.

ECHO PROFILE

This field is a folder link to the associated Echo Profile. See "Echo Profiles" on **page 735** for details.

SIP PHONE GROUP

(*For SIP phones only*) This field is a folder link to the associated SIP Phone Group. See **page 673** for details.

ASSOCIATED HOT DESK DEVICE

(*For Digital Telephones, 86xx IP phones, 52xx/53xx IP phones, and 8602 Softphones only*) This field links the physical phone with the Hot Desk Profile that is currently logged into it. When the Hot Desk Profile logs out of the Hot Desk session, this field is set to None. A user can only log on to a phone that is enabled for Hot Desking (see **page 447**).

NOTE	This field does not refresh itself automatically, but whenever it is re-displayed (due to folder change or refresh request from a DB Programming user) or whenever a Database Save is requested, it is re-queried to be current.

APPLICATION PROFILE

This field links the phone with an HTML Application Profile in System\Phone-Related Information\Application Profiles*<ID>*\Applications. The applications in the Application Profile are downloaded to the phone for use. If NONE is chosen, no HTML applications will be downloaded to that phone. It is set to 1 (default profile) by default. See **page 346** for details about Application Profiles.

PHONE/PHANTOM/HOT DESK PROFILE FLAGS

You can use flags to enable or disable phone features. Table 69 on **page 418** shows flags and descriptions. You can program flags for an individual device, or you can program flags for multiple phones.

PROGRAMMING FLAGS FOR INDIVIDUAL DEVICES

To program phone flags for an individual device:

1. Do one of the following:
   - Select – System – Devices and Feature Codes – **Phones**.
   - Select – System – Devices and Feature Codes – **Phantoms & Hot Desk Profiles**.

2. Select the extension number.

3. Select **Flags**.

4. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

5. Click out of the field or press **ENTER** to save your change. Some flags require you to choose between two settings, such as Hunt Group Remove/Replace. For those flags, select the current value and use the scroll box to select the desired value. Then click out of the field or press **ENTER** to save your change.

PROGRAMMING FLAGS FOR MULTIPLE PHONES

See Table 69 on **page 418** for phone flag descriptions.

To batch program flags for multiple phones:

1. Select System – Phone-Related Information – **Flags**.

2. Select the flag.

3. Right-click anywhere in the right side of the window (if there is no blank space, press **CTRL + N**). An option box appears.

4. Select **Add To *<name of flag>* List**. A window appears prompting for the device type to include. The choices are:
   - 52xx\53xx Phone
   - 86xx Phone
   - Digital Telephone
   - IP Single Line Adapter
   - IP Softphone
   - Phantom/Hot Desk Profile
   - SIP Phone
   - Single Line
   - UC Advanced Softphone

5. Select the device type, and then click **Next**. The devices with details appear. To view devices in a list only, click **List**.

6. Select the desired phones, and then click **Finish**. The selection appears in the appropriate field. To select a series of items, hold down **SHIFT** while selecting the first and last item in the range. To select two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

NOTE	Available flags may vary depending on the phone type.

Table 69: Phone/Phantom/Hot Desk Profile Flags

FLAG NAME	DESCRIPTION
Activate Door Relay	This flag determines whether the Activate Door Relay feature is allowed for individual phones. It is set to No by default. See "Door Relay Control" on **page 293** for details about Door Relay Control.
Administrator	A phone can be assigned as an administrator to permit use of the administrator feature codes at that phone. To assign the phone as an administrator phone, enable this flag.
All Transient Displays	When this flag is enabled, it allows all transient call handling displays to appear on the phone display. If it is disabled, the CALL TRANSFERRED TO XXX and MESSAGE LEFT FOR XXX displays does not appear. In the default state it is enabled.
Alternate Hold Timer	For users' convenience, the system has two Hold timers: *Hold* and *Hold - Alternate*. In the default state, the Alternate timer is set for a longer time period than the Hold timer. However, both timers are programmable. The "Alternate Hold Timer" phone flag determines which timer each phone uses. If the flag is disabled, the phone uses the Hold timer. If it is enabled, the phone uses the Alternate timer. In the default state, the flag is disabled.
Alternate Keymap	If an alternate keymap is used on the phone, it can be enabled through programming or by the phone user. This flag shows the current state of the alternate keymap (enabled or disabled). In the default state, the alternate keymap is *disabled* and the standard keymap is used. The phone user can switch keymaps at any time by entering a feature code.
Alternate Transient Display Timer	This flag, if enabled, allows the length of time that transient displays remain visible to be controlled by the "Digital/IP Alternate Transient Display" timer. If this flag is disabled, transient displays remain visible for 5 seconds. In the default state, it is *disabled.*

Page 1 of 5

Table 69: Phone/Phantom/Hot Desk Profile Flags (continued)

FLAG NAME	DESCRIPTION
Attendant	A phone can be designated as an attendant. It then appears in the list of available phones when you are assigning Associated Extensions for other phones. *If you attempt to remove attendant status from the primary attendant,* a warning message appears. The primary attendant can only be changed using the programming options described in "System Forwarding Paths" on **page 469**. Message centers and Attendant phones can serve phones located on other nodes. (See "Attendant Phone Features" on **page 423** for details about Attendant phones.) However, because there is not a common database for all nodes, the message center and attendant assignments for the off-node phones must be programmed using Associated Extensions for the individual phones on the remote nodes.
	Note that because the local node cannot see the databases of the other nodes, it is up to you to keep track of which attendants and message centers are serving off-node devices. Be careful not to remove those phones, leaving the off-node devices without an attendant or message center.
Audio Diagnostics	The Audio Diagnostics flag enables or disables the Audio Diagnostics feature for a phone. By default, this flag is enabled. To disable the Audio Diagnostics feature for the phone, set this flag to **No**. For a feature description, see "Audio Diagnostics" on **page 1113**.
Automatic Answer CO or Intercom	The Automatic Answer features can be enabled so that ringing outside and intercom calls are answered by simply lifting the handset. If automatic answer is disabled, the user must press a button to answer the call. This can also be programmed by the phone user. In the default state, automatic access for both IC and CO are *enabled.*
Caller ID [CLIP] CO or Intercom	*(For Phones only)* If the Caller ID [CLIP] IC or CO flags are ON and there are available transmitters on the DSPs, Caller ID [CLIP] information is sent to single line phones for intercom and outside calls. By default, these flags are OFF. For more information about Caller ID, DNIS, and ANI requirements for single line phones, see "Caller ID, DNIS, and ANI" on **page 172**.
Caller ID CO or Intercom (ports within the Base Chassis and the DEIs.)	*(For Phones only)* Caller ID is not available on SLAs or MDPMs. The Advanced CO Interfaces feature requires a software license. Single line phones can use ANI or Caller ID [CLID] to receive the calling party's number and name, if available, for outside and intercom calls. Because some single line devices require an extended ring cadence to receive Caller ID [CLID] information, you must first enable the Extended Ring Cadence flag to configure Caller ID [CLID] for IC calls. Without the Extended Ring Cadence set to Yes, the Caller ID [CLID] - IC flag cannot be set.
	If this flag is enabled, the single line phone will display the calling party's information. By default, this flag is disabled.
Camp-On Indications	The tone and display that signal a user when a call has camped on can be enabled or disabled. However, the lamp in the CALL or trunk buttons *cannot* be disabled. In the default state, the tones are *enabled.* Hunt group Camp On tones can be disabled without disabling phone Camp On tones. See also the Hunt Group Camp-On Audio Indications flag on **page 421**.
Camp-On to ARS	The phone can be permitted to camp on to a trunk when using Automatic Route Selection (ARS), or camp on to ARS can be disabled. In the default state, camp on to ARS is *enabled.*
CO/IC Resize	The ability to immediately resize a trunk or intercom channel, after disconnecting from a call, can be enabled. In the default state, it is *disabled.*

Table 69:   Phone/Phantom/Hot Desk Profile Flags (continued)

FLAG NAME	DESCRIPTION
DID/E&M [DDI in Europe] Receive Busy Instead Of Camp On	E&M and DID callers can receive a busy signal, or receive ringback and camp on, when calling a busy phone. If this feature is enabled, the caller hears busy signals when the called phone is busy, and does not camp on. In the default state, busy tones are *disabled* and callers hear ringback while camped on to the called phone.
Different Alerting Cadence Intercom/CO Call	To allow users to easily differentiate between ringing intercom and outside calls, separate cadences can be enabled. In the default state, the different cadences are *enabled.*
Different Ringback Cadence Intercom/CO Call	The ringback heard when a user places a call can be different for intercom and outside calls, or both types of calls can use the CO ringback cadence. In the default state, the different cadences are *enabled.*
Display Missed Calls On Phone	This flag is used for 53xx phone users that use Call History application. Because there are now two ways to view missed calls on 5320/30/40/60 phones, there are cases where accessing the missed calls lists will not synchronize the two views. This flag is used to mitigate issues with the "out-of-sync" missed calls information. It is set to Yes by default.
Display Outside Party Name	If this flag is enabled, the phone user can toggle the outside party name and number when connected to a CO call with outside party information. The phone user enters the Display Outside Party Name On/Off feature code (379). In addition, the enhanced ring-in displays will provide the user with more information such as both Caller ID [CLID] name and number if available, or tell the user if a Caller ID [CLID] number is blocked or out-of-area. This is a phone-only flag. In the default state, this flag is enabled.
Do-Not-Disturb Allowed	This field designates whether the phone can be placed in Do-Not-Disturb. In the default state, it is *enabled.*
Do-Not-Disturb Override	The phone can be given Do-Not-Disturb override privilege that allows the user to place an intercom call to a phone in Do-Not-Disturb. In the default state, it is *disabled.*
Do Not Propagate Original Caller ID to P-Asserted Identity	This flag allows the administrator to control whether or not the Original Caller ID will be propagated to the P-Asserted Identity header for SIP calls. Do Not Propagate Original Caller ID to P-Asserted Identity is only available if Propagate Original Caller ID is enabled.
Expanded CO Call Information On Displays	This flag determines whether call information (trunk name or caller information) is displayed at the phone. If it is enabled, the Outside Call Party Information Has Priority flag (described on **page 421**) determines what is displayed. If it is disabled, the programmed trunk username will appear on the display. In the default state, this flag is *enabled.*
Extended Ring Cadence	Extended ring cadence can be enabled or disabled. The extended setting lengthens the duration of the ring signal to meet the requirements of off-premise extension (OPX) and repeater applications. The default setting is *disabled.*
Handsfree On/Off	The phone can be programmed to receive intercom calls handsfree, or handsfree answering can be disabled. This can also be programmed at the phone. In the default state, handsfree answering is *enabled.* **IMPORTANT:** This flag must be disabled for all SIP phones.

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Table 69:   Phone/Phantom/Hot Desk Profile Flags (continued)

FLAG NAME	DESCRIPTION
Headset Connect Tone	Some headsets have a power-saver mode that disables the headphone after a period of silence. To prevent these headsets from missing portions of incoming calls, enabling this flag allows the phone to generate a tone that activates the headset before connecting to an incoming call. In the default state, it is disabled.
Headset On/Off	If a headset is used on the phone, it can be enabled through programming or by the phone user. In the default state, it is *disabled.*
Hot Desking Allowed (to this device)	Enables Hot Desking for this phone. You can toggle this flag any time, as long as Hot Desking is not active for the phone. When disabled, it will only prevent future Hot Desking to that phone. It is set to No by default. For details about Hot Desking, see **page 447**.
Hot Desking Allowed (from this device)	*(For Phones only)* Displays the list of phantom devices from which Hot Desking is allowed.
Hunt Group Remove/ Replace	The phone Hunt Group Remove/Replace feature can be programmed using this flag or using the feature code at the phone. In the default state, this feature is set for *replace,* which allows the phone to receive hunt group calls.
Hunt Group Camp-On Audio Indications	The tone that signals a hunt group member when a call has camped on to the hunt group can be enabled or disabled. (However, the camp-on display and lamp in the CALL or trunk buttons cannot be disabled.) In the default state, the tones are *enabled.* If all phone camp-on tones are disabled using the Camp-On Indications flag, hunt group camp-on tone is disabled regardless of the Hunt Group Camp-On Audio Indications flag setting.
Immediate OHVA Transmit	Determines if the phone has immediate Off-Hook Voice Announce capability enabled. If it is not enabled, the OHVA call does not go through until the Off-Hook Voice Announce Screening timer expires. In the default state, this field is *disabled*. Not supported in IP phones.
Manual Forward to Public Network	When disabled, this flag prevents the phone from manually forwarding calls to the public network. This prevents users from calling in to a local phone and reaching toll numbers via forwarding. In the default state it is *enabled*.
OHVA Receive/Transmit	(The OHVA Receive prompt does not apply to IP devices, phantoms, and Hot Desk profiles*.)* Any non-IP device can be programmed to place Off-Hook Voice Announce calls. Digital telephones can be programmed to receive Off-Hook Voice Announce calls. In the default state, receive and transmit are *disabled.* Note that the system-wide flag must also be enabled to use the feature.
OPX Gain Levels	*(For Phones only)* When a single-line circuit is used as an off-premises extension, this field can be enabled to set receive and transmit gain levels to better suit the OPX application. In the default state, OPX gain levels are *disabled* (the gain levels are set for on-premises use). *Although you can enable this flag for all single lines, the flag is ignored unless the single line set is a T1 OPX.*
Outside Party Call Information Has Priority	If the Expanded CO Call Information flag is enabled (see **page 417**), this flag determines what information is displayed at the phone. If enabled, any call that is received on a trunk that provides outside caller information, Caller ID (CLID) or ANI, will be identified on the phone display with the caller information. If disabled, the display will show the DID or DNIS information for the call (if available). In the default state, is it *enabled.*
Page Mode	Pages to the phone can be enabled or disabled through programming or by the phone user (page remove/replace). In the default state, pages are *enabled*.

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Table 69: Phone/Phantom/Hot Desk Profile Flags (continued)

FLAG NAME	DESCRIPTION
Propagate Original Caller ID on Transfer	When enabled, if the phone is on a call with an outside trunk that had caller ID information, when the phone performs a transfer back to the PSTN, the phone will propagate the caller ID. To propagate the caller ID to the PSTN, the eventual trunk must be ISDN and the "Propagate Original Caller ID" on **page 523** in its CO trunk group must be set to Yes. For details, "Caller ID Propagation" on **page 176**.
Receive Busy Instead of Do-Not-Disturb	The tone that a caller hears when a phone is in Do-Not-Disturb can be a busy signal or a Do-Not-Disturb signal. If this feature is enabled, the caller hears busy signals when the phone is in Do-Not-Disturb, but the caller cannot camp on. In the default state, busy tones are *disabled.*
Redial Mode	Phone users can use *dialed* or *saved* redial modes. This field can be programmed using this flag only, it cannot be changed by the phone user. In the default state, the phones are set for *dialed* mode.
Ring Intercom Always On/Off	When this feature is enabled, all calls placed to phones from this phone are received as private calls. The feature can be enabled or disabled through programming or by the phone user. In the default state, this feature is *disabled.*
Send Alert Burst To Headset	This flag, when enabled, sends a single tone to a headset instead of normal phone ring tones when a call is ringing at the phone (non-handsfree intercom call, outside call, queue callback, or reminder message). By default the flag is disabled.
Send T1 OPX Disconnect Flash	This flag, when enabled, sends a proprietary disconnect signal from the T1 OPX to the loop start trunk. (The "A" bit is toggled high for the duration of the SL Disconnect Flash Duration timer.) At default, this flag is *disabled.* Although you can enable this flag for all single lines, the flag is ignored unless the single-line set is a T1 OPX. This flag does not affect T1 channels configured for Loop Start. Therefore, a Mitel system can be on the receiving end with T1 Loop Start channels and will recognize the disconnect. However, it cannot **send** the disconnect. See "T1 OPX Service" on **page 189** for details about T1 OPX.
Standard Tone Frequencies	The busy tones and dial tones can be pure system tones or they can be changed to more closely match the service provider standard tones. In the default state, standard tones are *enabled*, which allows standard service provider-type tones.
System Forwarding	When this field is enabled, calls to the phone follow the system forwarding path that is programmed for the phone (if any). This field can be enabled is the database or at the phone. In the default state, it is *enabled.*
Transfer-To-Connect Allowed	If the Transfer-To-Connect Allowed flag is enabled, an announced transfer call (including an announced transfer from voice mail) connects immediately with the destination once the transferring party hangs up. (If the flag is disabled, the recipient must press a CALL button to answer the transferred call).
	If the user has a headset enabled, or if the phone user picks up the handset for the initial call and the transferring party disconnects, the transferred party hears a single tone and is then immediately connected without any further action from the phone user. Calls do not automatically connect in handsfree speakerphone mode; the user must press a CALL button to answer the call. This flag is *disabled* by default.
Transient Call Indication On Call Answer	This flag determines whether the phone user sees a call indication display when answering a call by pressing a secondary extension button or by reverse transferring. If enabled, the display indicates if the call was ringing, recalling, transferred, or holding at the other phone. In the default state, it is *enabled.*

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ATTENDANT PHONE FEATURES

This section describes features available to local and network Attendants through Attendant-assigned phones.

ATTENDANT PHONES

Attendant phones can be called by dialing 0 at the phones they serve. They are programmed to provide these services:

• Central operators for incoming calls

• Message centers

• Recall phones for unanswered calls

Any phone can be designated as an Attendant during database programming. Attendant locations should be equipped with display phones to show recall sources.

A hunt group can be assigned to serve as an attendant. However, the individual phones in the hunt group are not required to be programmed as attendant phones, and the database will not reflect that the individual phones serve as an attendant.

For programming instructions, see "Voice Processor Features and Programming" on **page 835**.

NETWORK AND LOCAL PRIMARY ATTENDANTS

One attendant can be designated as the primary attendant who can receive unsupervised outside call recalls, hunt group recalls, and calls that cannot be matched to patterns in call routing tables.

When installed in a network, a telecommunications system can support two types of primary attendants:

• **Node Attendants**: Each node can support a primary attendant.

• **Network Primary Attendant**: Each node can support a network primary attendant, but typically all of the nodes share one Network Primary Attendant. When the network needs to direct a call to an attendant, it attempts to direct the call to the network primary attendant first. If the network primary attendant is unavailable, it directs the call to the attendant on the node where the call originated.

If there is no network or local primary attendant, calls that would normally go to the primary attendant are handled as follows:

• If the system has seized the call, but it has not been sent to a phone, the call is disconnected.

• If the call has been sent to a phone, it remains at the phone location and rings until answered.

• If the call is not seized and not sent to a phone, the caller will hear ringing until he or she hangs up. The call will not ring at any phone location.

SINGLE-ATTENDANT OR MULTIPLE-ATTENDANT OPERATION

Each node can be set up with one attendant or several attendants, as follows:

- **One Attendant**: One attendant provides all of the services listed in **page 423**. In a typical system configuration, all trunks (except private trunks) are programmed to ring at this attendant's phone.

- **Multiple Attendants**: Any or all phones can be programmed as attendants. For example, one or more attendants might serve different departments in a business environment. Trunks are programmed to ring at any or all attendant phones. Multiple attendants can be arranged in a hierarchy. That is, one attendant may be the attendant for another. In this case, the "serving" attendant is reached by dialing "0" at the "served" attendant.Hunt Group Features.

KEYMAPS

You can view and program phone and DSS/PKM keymaps.

The Default keymap contains the default key programming for all IP/digital telephone models. You cannot delete or edit key assignments in the Default keymap. All new keymaps created in DB Programming have a copy of the Default keymap keys.

For databases converted or upgraded to a new DB Programming version, existing keymaps are preserved, but their IDs are incremented by 1. Existing keymaps are assigned a Default keymap type, and changes to this field are saved. Existing phones retain their keymap association, and any new phones are created with reference to the Default keymap. For more information about the Database Converter Utility, refer to the DB Converter Utility Help.

In a default database, the Default keymap is the only keymap programmed. Phones in the default database and any new phones are created with reference to the Default keymap.

The figure below shows the main IP/Digital Telephone keymap list for the default database.

For the Default keymap, the Description, Keymap Type, and User Programmable Keys are read-only. However, you can move IP/digital telephones to and from the Standard and Alternate Lists.

This section contains the following information:

- Viewing Default Keymaps on **page 426**

- Programming Keymaps on **page 426**

VIEWING DEFAULT KEYMAPS

The Default keymap contains the default key programming for all IP/digital telephone models. You cannot delete or edit key assignments in the Default keymap. All new keymaps created in DB Programming have a copy of the Default keymap keys.

To view a default phone keymap:

1. Do one of the following:

   - *To view the default keymap available to a particular phone group:* Select System – Phone-Related Information – Key Assignments – *<phone type>* – **Default**.

   - *To view the keymap assigned to a particular phone:* Select System – Devices and Feature Codes – Phones – (Local) – *<extension>* – **Keymaps**.

2. In the **Keymap** column, double-click **Default**.

3. Double-click **Keymap Type**. The selected keymap appears, as shown in the following example.
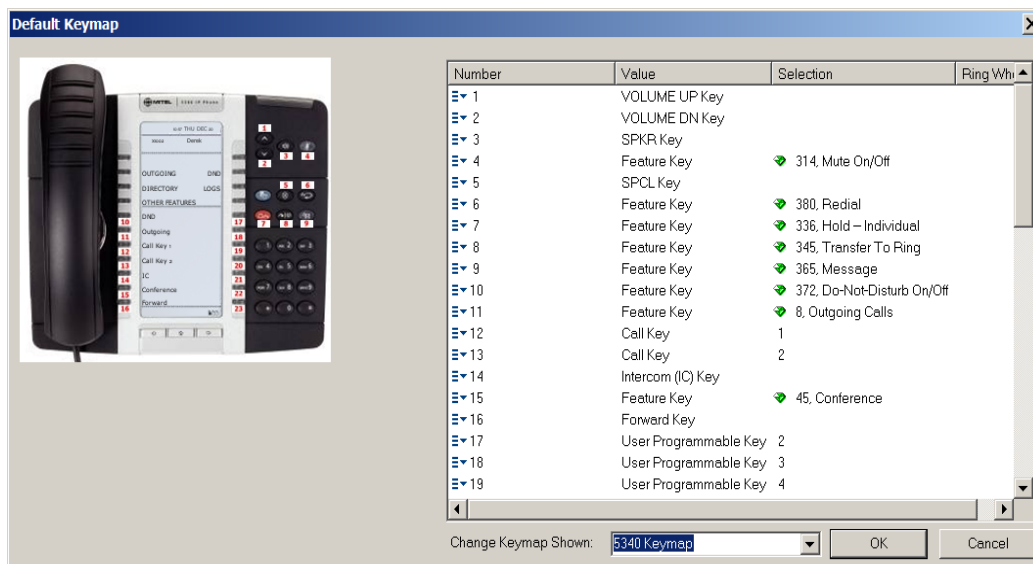
4. From the **Change Keymap Shown** list, select the phone type that you want to view. The phone image with key assignments appears. The phone image is view only—there are no operable controls in the image. See "Keymap Number Column" on **page 427** for more information about the Number, Value, Selection, and Ring When columns shown in the right pane.

PROGRAMMING KEYMAPS

This section contains the following information:

- **Adding New Keymaps** below

- Programming Phone Keymaps on **page 427**

- Selecting Standard or Alternate Keymaps on **page 435**

- Changing Keymap Types on **page 437**

- Copying and Pasting Keymaps on **page 438**

- Programming Phone Keymap Buttons on **page 439**

- Programming DSS/PKM Keymaps on **page 440**

ADDING NEW KEYMAPS

You can add new keymaps and assign them as standard or alternate phone keymaps to a particular phone type.

To add a keymap to the IP/Digital Telephone List:

1. Select System – Phone-Related Information – Key Assignments – **IP/Digital Telephone**.

2. Right click in the right pane, and then click **Add to IP/Digital Telephone List**. The Get Keymap IP dialog box appears.

3. From the Starting ID list, select the starting ID (2-26) and the number of IDs. You can create up to 25 keymaps.

4. From the Number of IDs list, select the number of keymaps you are adding.

5. From the **Keymap Type** list, select a keymap type.

6. Click **OK** to save the changes.

PROGRAMMING PHONE KEYMAPS

Remember the following when programming keymaps:

- Keymaps can only be used for the phone type and model for which they are programmed. For example, you cannot use a digital telephone keymap for an IP phone, and you cannot use an 86xx phone keymap for a 52xx\53xx phone.

- Key assignments within a keymap are common to all phone types in that map. Modifying a button changes that button assignment for *all* phones assigned to that map, regardless of phone type. For example, if the top left button is changed to a user-programmable button for six-line display and Executive Display phones in Keyset Map Group 1, the corresponding button is also user-programmable on all other phones in Keyset Map Group 1.

- Users who have Standard and Alternate keymaps want to have their user-programmable buttons in the same positions in both keymaps. That way their user-programmable button locations do not change when they switch keymaps. Also, if a fixed feature button is created for the Switch Keymap feature code, make sure it is in the same position in both keymaps.

KEYMAP NUMBER COLUMN

The keymap number shown in the right pane is the number that corresponds to the red number in the image. It is not programmable. It simply associates the list control item with the key.

KEYMAP VALUE COLUMN

The keymap value shown in the right pane is the current key type assigned to the key. You can program the keymap value.

To change the key type:

1. Click the **Value** field. A list box appears as shown below.

2. Click again to see the list of choices, as shown at right. The available values in the list varies depending on the phone type.
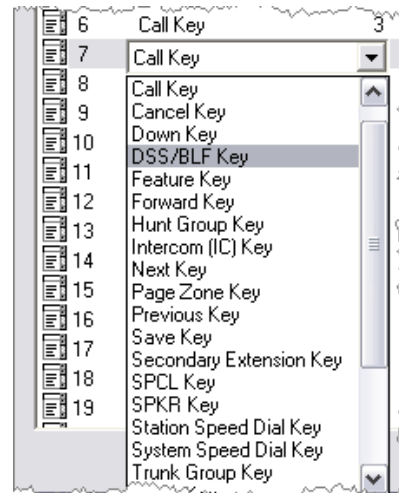
3. Click the new key type, and then click **OK**. The new key type appears. When you change the value, the Selection and Ring When columns are reset, depending on the new value. For a System Speed Dial Key, the Selection is set to the first System Speed Dial Number. For a Hunt Group Key, it is set to the first hunt group. The Ring When column is inoperable for all key types (values) except the Secondary Extension. For more information about key types, see "Associated Extensions" on .

KEYMAP SELECTION COLUMN

The Selection column includes additional information for the following key types:

- Call Key below

- User Programmable Key below

- Station Speed Dial Key on

- System Speed Dial Key on

- Device Key on

For all other key types, the Selection column is blank and inoperable.

- **Call Key**: A call key number is shown. This is assigned automatically and is not programmable. Call keys are always numbered sequentially in the order created. If one is removed, the gap is closed by renumbering those with higher numbers. A new Call Key is always assigned the next available number.

  To the end user, the first non-IC call is associated with Call Key 1. If another non-IC call is made or received while the first call is still active, the second call is associated with Call Key 2. The next call is associated with Call Key 3, and so forth. As soon as the first non-IC call is completed, Call Key 1 becomes available and is associated with the next non-IC call.

Because the Call Keys are used in the manner described above, if the key assignments are changed, create any new Call Keys in a thoughtful order, such that the numbering flows with the key positions.

- **User Programmable Key**: A key number is shown. The key is programmable, with a range of 1–45, and it always defaults to 1 when a new User Programmable Key is created. (You can program two User Programmable Keys with the same number, but this is not recommended.) The numbers correspond to the numbers that appear in two other folders: the "Phone User Programmable Keys" folder and the keymap "User Programmable Keys" folder.

  - In the System\Devices and Feature Codes\Phones\<*extension*>**Programmable Keys** folder, User Programmable Keys 1–45 can be programmed differently for each individual phone. These keys can also be programmed from the phone using a feature code. User Programmable Keys must be programmed and assigned with numbers in the keymap associated with the phone. The number assignment, 1–45, allows the key assignment made in the keymap to be associated with the programming in the Phone User Programmable Keys Folder.

  - In the System\Phone-Related Information\Key Assignments\IP/Digital Telephone\<*keymap ID*>**User Programmable Keys** folder, defaults for these keys can be programmed. Default programming is included in the Default keymap, and changes can be made after keymaps are created. The programming in this folder is associated with the keymap. For phones associated with the keymap, the programming for those keys can be copied to the phone. This capability allows for newly created phones to get the default programming for these keys. It also allows for the programming for these keys at the phone to be reset to match the keymap. (This is accomplished through the shortcut menu in a Keymap folder or in the Standard List or Alternate List folder of a keymap.)

  To edit the User Programmable Key number:

  1. Click the number in the Selection column. An edit box appears, as shown below.

  2. Enter the new number (1–45), and then click **OK**.

- **Station Speed Dial Key**: This number corresponds to the digit on the phone that is used to store or access this Station Speed Dial number. You can store up to 10 Station Speed Dial numbers for each phone, using digits 0–9. Programming a key to be a Station Speed Dial Key provides for station speed dial through a single key press. (Otherwise, the user dials the Station Speed Dial feature code, and then presses a digit 0–9 to indicate which Station Speed Dial Number to dial.) Station Speed Dial number programming must be done separately, using DB Programming or at the phone itself, by the Program Station Speed Dial feature code.

  This number defaults to 0 when a Station Speed Dial key is created. You can program two Station Speed Dial Keys with the same number, but this is not recommended.

To edit the number:

1. Click the number in the **Selection** column.
2. Type or select the new Station Speed Dial number, and then click **OK**.

- **System Speed Dial Key**: A System Speed Dial ID is shown, along with the programmed number, if any. When a new System Speed Dial Key is created, the ID is set to **0**. You can program two System Speed Dial Keys with the same ID, but it is not recommended.

  There are three ways to edit the ID: (1) type the ID manually, (2) use a Selection Box, or (3) use a Selection Wizard. These methods are explained in the following pages.

To type the ID manually:

1. Click the **Selection** column that you want to edit. An edit box appears, as shown below.

2. Type the ID (000–999), and then click **OK**. If the ID entered corresponds to an existing System Speed Dial Number, when you click away and then click **OK**, the change is completed.

   If the ID entered does not correspond to an existing System Speed Dial Number, an empty Select an Item box appears, as shown below.

3. Click **Cancel**, and then try again. The title of this box has changed from "Select a Device" to "Select an Item" to describe the functionality better.

To use a Selection Box:

1. Click the **Selection** column that you want to edit. An edit box appears, as shown below.

2. Click **OK**. When the edit box is empty, the Select an Item box appears showing all of the available System Speed Dial numbers for the key.
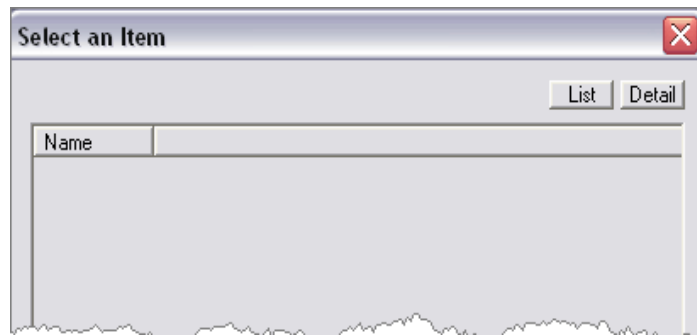
3. Select the number that you want, and then click **OK**.

To use a Selection Wizard:

1. Right-click the **System Speed Dial Key**, and then select **Change Selection**.
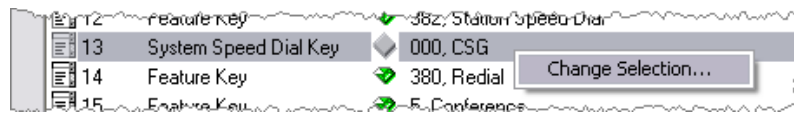
A Selection Wizard appears, presenting the available types of items that can be associated with the key.
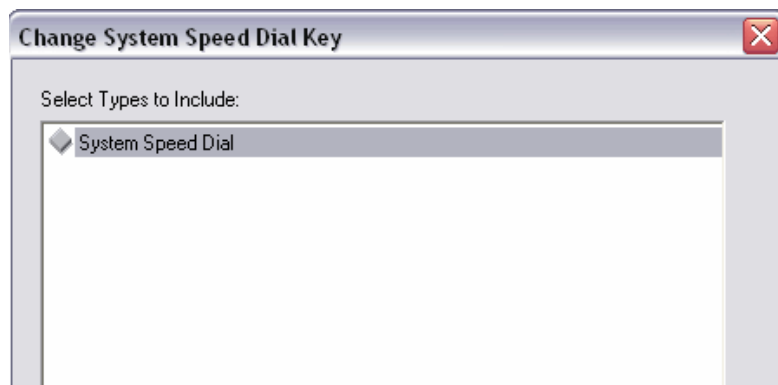
2. Click **Next** to accept the only choice of System Speed Dial. All of the available System Speed Dial numbers for the field appear in the box.
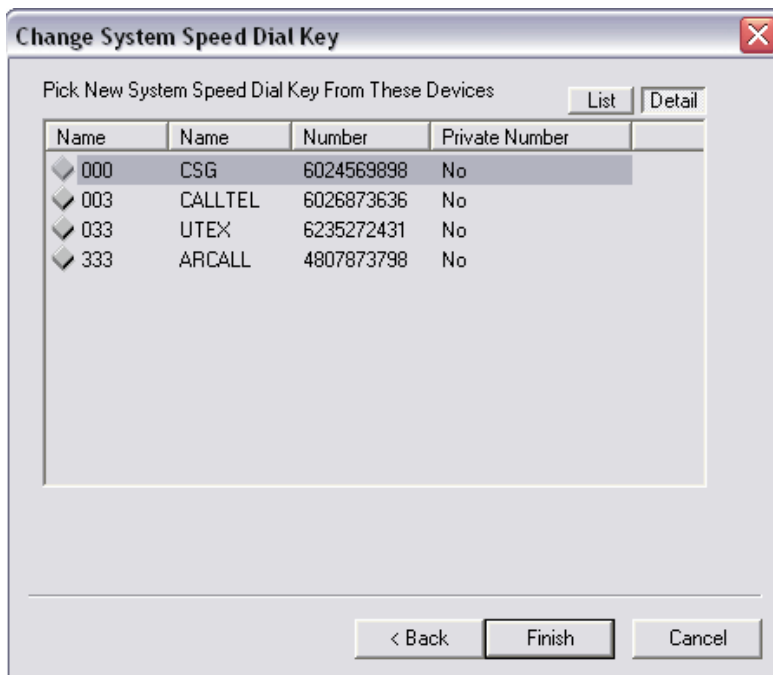
3. Click the **System Speed Dial Number** that you want to edit, and click **Finish**. If a new System Speed Dial Number was selected, it appears in the Selection column in the list control. Otherwise, the original number remains.

- **Device Key**: A "Device" key is any key type that is associated with a device (for example, Hunt Group Key, Trunk Key, Page Zone Key, and so forth) For a "Device" key, the Selection column contains the extension and description (if any) of the associated device.

When a key of one of these types is created, a default choice is always stored in the Selection column. This is always the first device found in the database appropriate for the type of key. For example, for a Feature Key, it is 0, Call Attendant. For a Page Zone Key, it is the first Page Zone. For a Trunk Group Key, it is the first CO Trunk Group, and so forth.

There are three ways to edit the ID: (1) type the extension manually, (2) use a Selection Box, or (3) use a Selection Wizard. These methods are explained below.

To type the extension manually:

1. Click the **Selection** column that you want to edit. An edit box appears, as shown below.

2. Type the extension that you want, and then click **OK**. If a complete, valid extension is entered, the change is completed, the Popup Edit Box is removed, and the extension is shown, along with the device description (if any).

3. If the extension is incomplete or invalid, the Select an Item box appears as shown below.
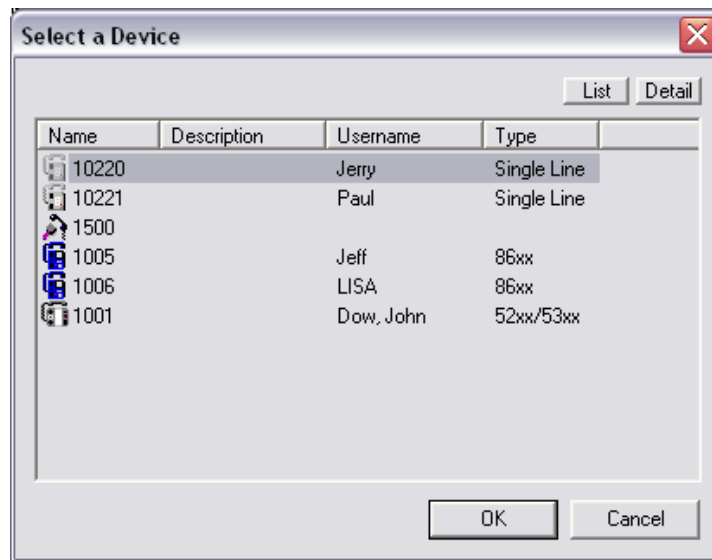
For an incomplete extension, the box contains only the extensions of devices appropriate for the key that begins with the digits entered. For example, if you are editing a DSS/BLF (PKM) Key Selection and enter "1" and then click **OK**, the Selection Box contains only devices with extensions beginning with "1" that can be placed under a DSS/BLF (PKM) Key. You can then find and select the device that you want, and then click **OK**.

If the entered digits do not correspond to the beginning digits of any devices of the type(s) associated with the key type, the box clears. Click **Cancel**, and then try again.

To use a Selection Box:

1. Click the **Selection** column that you want to edit. An edit box appears, as shown below.

2. Click **OK**. When the edit box is empty, a Select an Item box appears showing all of the available phones, voice mail applications, hunt groups, and so forth. For a Feature Key, all of the feature codes are shown. For a Hunt Group Key, just the hunt groups are shown, and so forth.
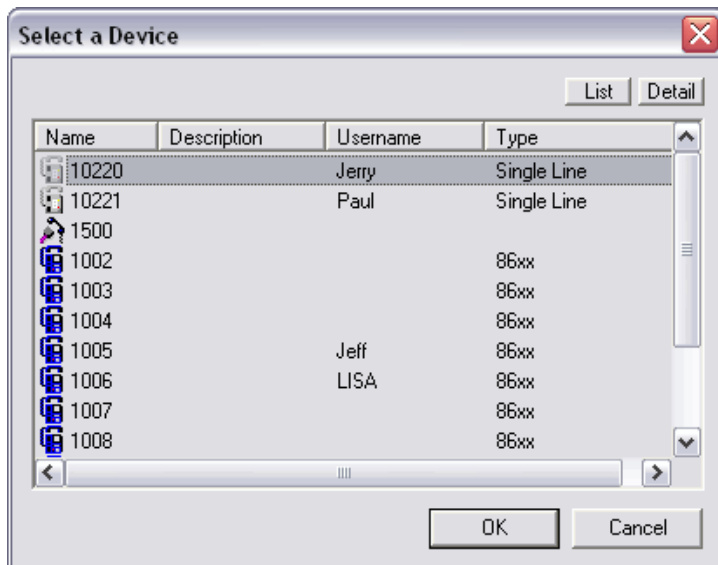
3. Select the device that you want, and then click **OK**.

To use a Selection Wizard:

1. Right-click the **DSS/BLF (PKM) Key**, and then select **Change Selection**.

A Selection Wizard appears, presenting the available types of items that can be associated with the key.

Select the device types, and then click **Next**. A list of selected device types appears in the box. The example below illustrates when the 86xx Phone and IP Softphone types are selected.
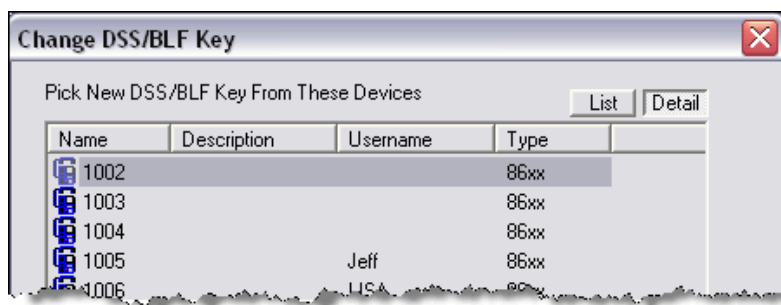
2. Select the device you want, and then click **Finish**.

- **Ring When**: This field is blank and inoperable for all key types except for a Secondary Extension Key. When a key of that type is created, the Ring When column defaults to 1. It can be set to any value from 0 to 10. The value indicates how many additional calls need to be ringing at the phone before the assigned secondary extension begins to ring. Setting the value to 0 turns off the ring indicator for the programmed key.

To edit this column:

1. Click the number in the **Ring When** column. An edit box appears, as shown below.

	21	User Programmable Key	4	
	22	Secondary Extension Key	10220	1
	23	User Programmable Key	6	
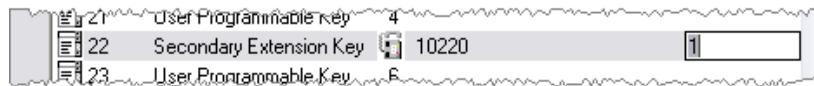
2. Edit the number as needed, and then click **OK**. Only digits (up to 10) are accepted in this box. If the number typed is out of range, the following message box appears, and the change is ignored.

> **Database Programming** ☒
>
> ⚠ The value specified is invalid. The range of allowable values is 0 through 10.
>
> [OK]
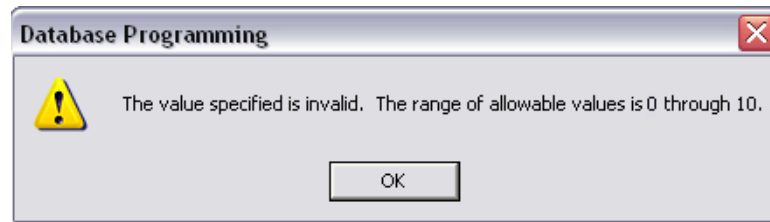
SELECTING STANDARD OR ALTERNATE KEYMAPS

All digital and IP phones created in DB Programming are assigned to the Default keymap. To associate a phone with another keymap, either browse to the Keymap subfolder of the phone, or browse to the Standard or Alternate List subfolder of the keymap that you want to edit and move the phones to this folder using the shortcut menu.

Because DB Programming differentiates between 52xx\53xx phones and 86xx phones, only 52xx\53xx phones may be moved to a keymap that is programmed for a 52xx\53xx phone keymap type. Likewise, only 86xx phones or digital telephones may be moved to a keymap programmed for an 86xx phone keymap type. When the keymap type is changed, either by moving phones from one list to another or by changing the keymap reference in the phone subfolder, the system prompts you to default the Programmable Keys of the phone to match the User Programmable Keys of the keymap. If you click **No**, the Programmable Keys of the phone(s) remain the same. If you click **Yes**, the User Programmable Keys of the keymap that the phone has been changed are copied over the Programmable Keys of the phone(s). This is the same functionality as the shortcut menu item "Default Selected User Keys" in the Standard/Alternate List folders.

To select a keymap for a phone:

1. Select System – Devices and Feature Codes – Phones – (Local) – *<extension>* – **Keymaps**.

2. Right-click either Standard or Alternate IP/Digital Telephone Keymap Group for the map you want to change, and then click **Change Standard/Alternate Digital Telephone Keymap Group**. The Change Standard/Alternate IP/Digital Telephone Keymap Group dialog box similar to the one below appears, prompting you to select the type of keymap to include.

3. Select the appropriate keymap, and then click **Next**. The list of keymaps appears. To view keymap details, click **Detail**.
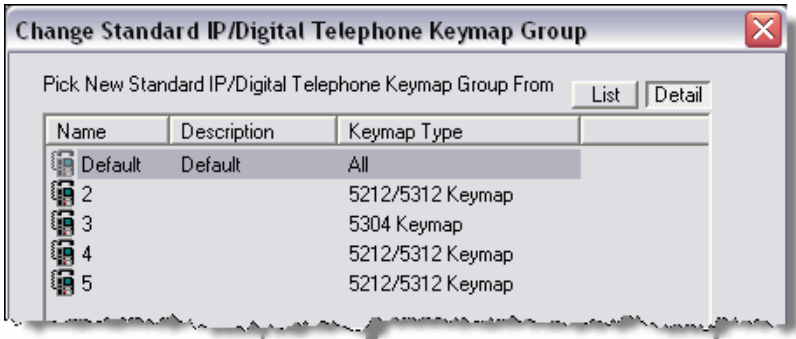
4. Highlight the keymap that you want to use, and then click **Finish**. The selected keymap appears in the Keymap column.
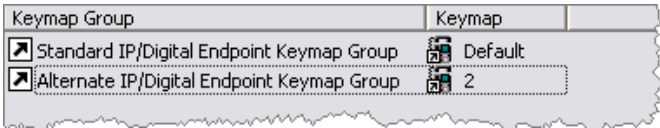
CHANGING KEYMAP TYPES

You cannot change default keymap types. When you change a keymap, the keymap key assignments themselves do **not** change.

Each keymap must have an associated "Keymap Type." The Keymap Type column is in the IP/Digital Telephone Keymap list and the individual IP/Digital Telephone keymap folder. You can program the keymap type from either folder, as shown in the figure below.

Shown in IP/Digital Telephone Keymap list
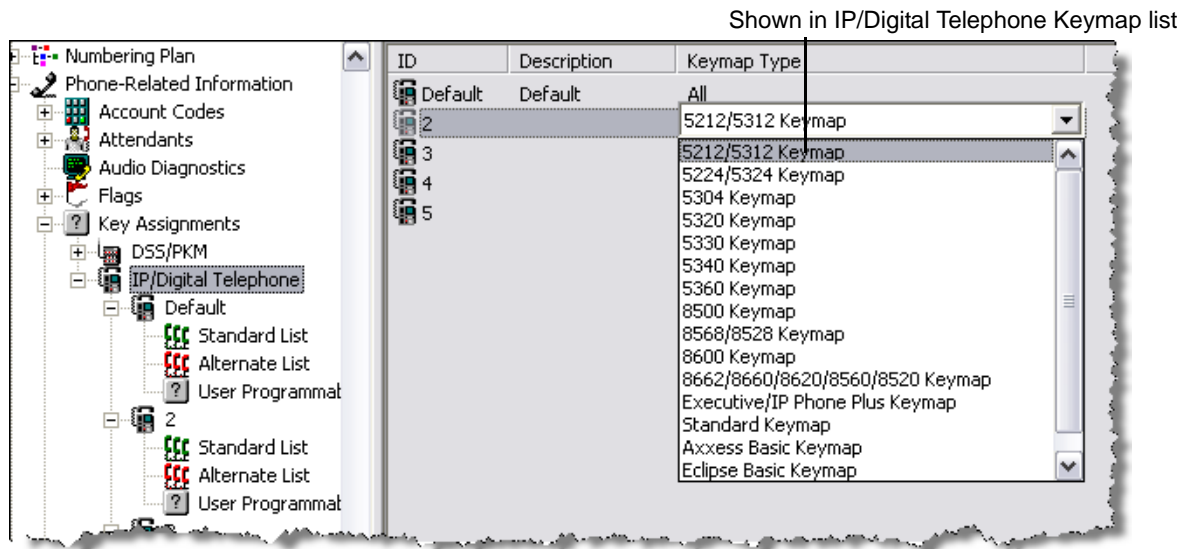
Shown in individual IP/Digital IP/Digital Telephone keymap
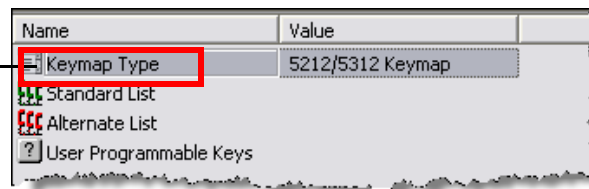
To change the Keymap type:

1. Do one of the following:

- Select System – Phone-Related Information – Key Assignments – IP/Digital Telephone – <**keymap**>.

- Select System – Devices and Feature Codes – Phones – (Local) – *<phone>* – Keymaps – **Alternate** (or **Standard**) **IP/Digital Telephone Keymap Group**.

2. Select the keymap box, which is either the Keymap Type column field in the main IP/Digital Telephone keymap list or the Value column field in the individual IP/Digital Telephone keymap folder.

3. From the **Keymap Type** list, select the keymap type. If the keymap type is changed from a 52xx/53xx model to an 86xx model or vice versa, a warning message appears indicating that any phones referencing that keymap will reference the Default keymap instead. Click **Yes** to continue.

SIP Peer Trunks Limitations

SIP peer trunks have limited functionality compared to other trunks. If a user programs a key as a SIP peer trunk, the key has the following limitations:

- The LED status for a SIP Peer Trunk key does not reflect the status of the individual SIP peer trunk.

- The LED status for a SIP Peer Trunk key reflects the status of the entire SIP peer trunk group.

- SIP peer trunks do not support the system hold feature.

For complete information about SIP peer trunks and trunk groups, see "Service Provider SIP Trunks and SIP Trunk Groups" on **page 691**.

COPYING AND PASTING KEYMAPS

You can copy and paste keymaps. Remember the following when copying and pasting keymaps:

- The Description and the Standard and Alternate Lists are not copied or pasted with a keymap.

- The Default keymap is not a valid paste location. Attempting to paste to the Default keymap displays an error message.

- A keymap for an 86xx keymap type may not be copied or pasted to a keymap with a keymap type for a 52xx\53xx Phone model and vice versa. If you attempt to copy or paste a different model keymap, an error message appears.

To copy and paste the entire keymap:

1. Right-click the keymap that you want to copy, and then select **Copy**. When selected, the database copies the keymap type and key assignments (including the User Programmable Keys) from that keymap to the Clipboard.

2. Select the keymap to which you are pasting.

3. Right-click, and then select **Paste**. The Paste option is available only when the Clipboard holds a copy of a keymap and you right-click on a keymap in the Keymaps folder. If a group of keymaps is selected, and you right-click in the group, the type and key assignments are pasted into each selected keymap. Otherwise, the type and key assignments are pasted into the single keymap.

   If you attempt to paste a keymap of one type onto a keymap of another type, a confirmation message appears. Click **Yes** to confirm.

To return a keymap to its default key assignments:

1. Right-click the keymap that you want to modify, and then select **Default**.

2. When the confirmation message appears, click **Yes** to default both the key assignments and the User Programmable Keys. Click **No** if defaulting only the key assignments. Note that the Description, Keymap Type, and Standard and Alternate Lists will not be changed.

PROGRAMMING PHONE KEYMAP BUTTONS

All Programmable Key lists have 45 keys. Keys 36–45, as shown in the figure below, are used for 52xx\53xx phones only.

52xx\53xx phones only
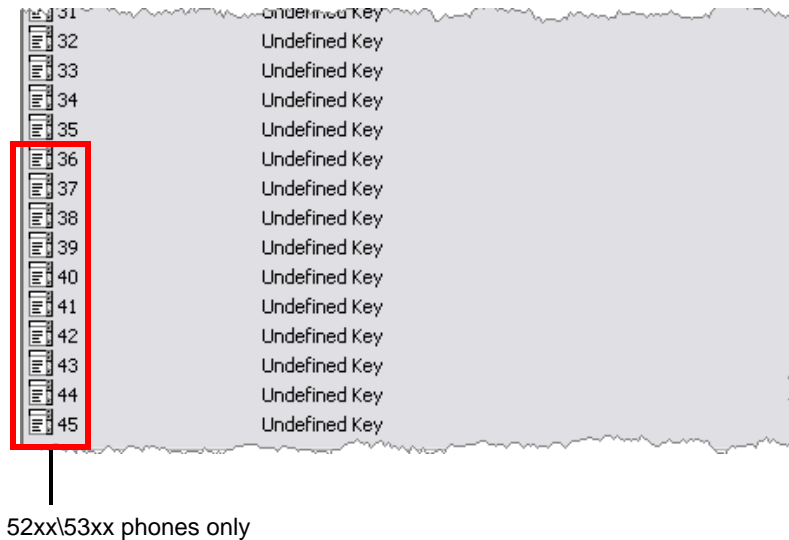
You can change the programmable keys assigned to a keymap for all phones assigned to the keymap, or you can edit the programmable keys for a particular phone.

To program phone keys:

1. Do one of the following:

    To program keys for a keymap:

    Select – System – Phone-Related Information – Key Assignments – IP/Digital Telephone – *<phone keymap>* – **User Programmable Keys**.

    To program keys for an individual phone:

    Select – System – Devices and Feature Codes – Phones – (Local) – *<extension>* – **Programmable Keys**.

2. *To change the Key Type, Selection, or Ring When options*, do the following:
    *   **Key Type**: Select the current Key Type for the button and scroll through the drop-down list box until the desired key type is selected.
    *   **Selection**: The Selection options required for each of the key types are as follows:
        *   **DSS/BLF (PKM) Key**: Select the circuit or extension number of the station that will appear under the selected DSS/BLF (PKM) button.
        *   **Feature Key**: Enter the feature code that is activated when the button is pressed.
        *   **Hunt Group Key**: Select the pilot number of the desired hunt group.
        *   **Intercom (IC) Key**: No further programming is necessary. This button is used for seizing an intercom channel or answering a waiting intercom call.

- **Page Zone Key**: Select the page zone 0–9 that will be used by selecting the page zone access code 9600–9609.

- **Secondary Extension Key**: Select the primary phone that will be associated with this secondary extension button.

- **Station Speed Dial Key**: Select the Station Speed Dial location (0–9) that will be assigned.

- **System Speed Dial Key**: Select the System Speed Dial location (000–999) that will be assigned.

- **Trunk Key**: Select the individual trunk that will be selected when this button is pressed.

- **Trunk Group Key**: Select the trunk group that will be selected when this button is pressed.

- **Undefined Key**: No further programming is necessary. This button can be programmed by the user.

To program the selection, right-click the Selection value, and then do the following:

1. Select **Change Selection**. A window appears prompting for the type to include.
2. Select the type (feature code, phone, and so forth), and then click **Next**. The items with details appear. To view items in a list only, click **List**.
3. Select the appropriate item, and then click **Finish**. The selection appears in the Selection field.

- **Ring When**: (Secondary Extensions only). Allows the phone to receive a burst of ringing when a certain number of calls are present at a primary phone, and at least one call is ringing or camped on. The ring burst repeats periodically as long as the programmed number of calls are present at a primary phone. The period of repetition is determined by the Digital/IP Secondary Extension Key Alerting Tone timer. If this field is set to 0, the secondary phone never rings.

To set this number:

1. Select the current Value and scroll to or enter the new value in the box.
2. Click out of the field or press **ENTER** to save your change.

PROGRAMMING DSS/PKM KEYMAPS

When you double-click a specific DSS/PKM Keymap, you have the following options:

- **DSS/PKM Map**: The following DSS/PKM options are available:

  - PKM 12

  - PKM 48

  - *PKM 50*—corresponds to the 50-key DSS Unit (previously known as the Model 8450 DSS).

  - *60-key DSS Keymap 6-row and 12-row Model options*—correspond to the 60-key DSS Unit.

> **NOTE** To program a PKM 16 (mini-DSS), you must use the User Programmable Keys on a phone (see **page 429**).

- **DSS/PKM Phone List**: When you double-click **Phone List**, you see the phones that are currently assigned to the DSS/PKM Keymap. See "Programming DSS/PKM Phone Lists" on **page 444**.

You can automatically or manually populate a keymap, as described in the following sections.

AUTOMATICALLY POPULATING DSS/PKM KEYMAPS

You can automatically populate DSS/PKM Keymaps with system phones.

To automatically populate the DSS/PKM or keymap:

1. Do one of the following:
   - Select System – Phone-Related Information – Key Assignments – DSS/PKM – *<DSS/ PKM number>*.
   - Select System – Devices and Feature Codes – Phones – (Local) – *<extension>* – Keymaps – DSS/PKM Keymap Group – **DSS/PKM Keymap**.

2. Right-click the DSS/PKM keymap, and then select **Populate DSS/PKM Keymap**. A message appears notifying you that DB Programming will automatically populate the keymap with the system phones. To do this, the system starts at the lowest extension value and populates every button in ascending order. If there are more DSS/PKM buttons than extensions, any buttons not assigned an extension are changed to "Undefined." If you populate the keymap when there are no extensions available, all buttons are changed to "Undefined."

MANUALLY POPULATING DSS/PKM KEYMAPS

You can populate the DSS/PKM keymap with selected devices.

To manually populate the keymap:

1. Do one of the following:
   - Select System – Phone-Related Information – Key Assignments – DSS/PKM – *<DSS/ PKM number>*.
   - Select System – Devices and Feature Codes – Phones – Local – <extension> – Keymaps – DSS/PKM Keymap Group – **DSS/PKM Keymap**.

2. Double-click **DSS/PKM Keymap** to view an illustration of the device, as shown in the following example. DSS/PKM button types are described in Table 70 on **page 443**.
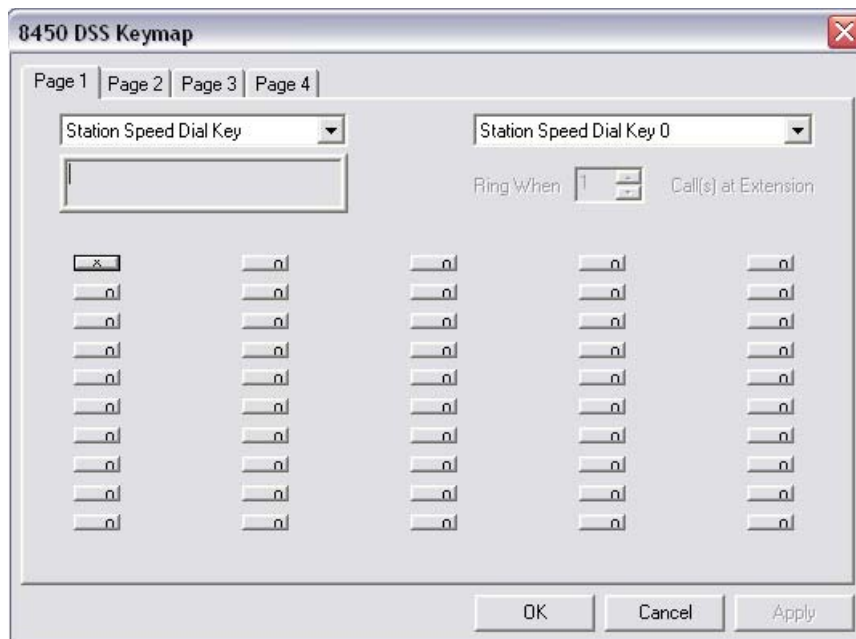
The DSS/PKM Keymap shows four pages of buttons—one for each device that can be connected to the phone. To see the buttons on a particular page, select the tab at the top of the page.

3. Do one of the following:

• *If you are changing a button requiring additional information, such as a feature code, extension number, and so forth*:

a. Click the button for which you want to assign a new value. A list window appears.

b. Select the device type from list, and then select **Next**.

c. Highlight the new value from the available list, and then select **Finish**. The new value appears in the box below the Key Type box.

• *If you are changing a button that does not require additional information:* Use the arrow keys on your keyboard to move around the keymap. Once you have selected the desired button (selected buttons will have a line with an "x" in the center), click **Key Type** to scroll to the new key type. If you change the button type to one requiring additional assignments (for example, extension, hunt group, and so forth), any previously programmed information for that button type remains. To change this information, you must click the button again and follow the directions detailed previously.

NOTE	Secondary extension buttons also have an additional option, "Ring When _ Calls At Extension. This option allows the station to receive a burst of ringing when a certain number of calls are present at a primary station, and at least one call is ringing or camped on. If this option is set to 0, **the secondary station will never receive the burst of ringing**. To set this number, select the current value and scroll to or enter the new value in the box. Click out of the field or press **ENTER** to save your change.

Types of keys, or buttons, and the selections that are required are described in **Table 70**.

Table 70:   DSS/PKM Button Types and Descriptions

KEY	DESCRIPTION
Accept Key	No further programming is necessary. This button is used to accept an entry when using numeric mode. This button type is *not* required for digital telephones. On digital telephones you activate the VOLUME function by pressing both arrows at once
Call Key	The next CALL button number is automatically selected.
Cancel Key	No further programming is necessary. This button will be used to cancel a feature or cancel an entry when in numeric mode.
Down Key	No further programming is necessary. This button can be used to scroll backward through displays. When the phone is idle, it can be used for adjusting volume, in place of the VOLUME button. This button type is *not* required for digital telephones. On digital telephones you use the VOLUME DOWN button.
DSS/BLF (PKM) Key	Select the circuit or extension number of the phone that appears under the selected DSS/BLF (PKM) button. If this is assigned to an off-node device, the button can be used for dialing the phone, and the lamp will show phone status.
Feature Key	Select the feature code that will be entered when this button is pressed.
Forward Key	Select the desired Call Forward feature code.
Hunt Group Key	Select the pilot number of the desired hunt group.
Intercom (IC) Key	No further programming is necessary. This button will be used for seizing an intercom channel or answering a waiting intercom call.
Next Key	This button type is ***not*** required for digital telephones. No further programming is necessary.
Page Zone Key	Select the page zone 0–9 that will be used by selecting the page zone access code 9600–9609.
Previous Key	(This button type is *not* required for digital telephones.) No further programming is necessary.
Programmable Key	Select the desired programmable button number (1–20).
Save Key	No further programming is necessary. This button type is *not* required for digital telephones. Does not apply to 52xx\53xx phones, because volume changes made on 52xx\53xx phones are automatically saved.
Secondary Extension Key	Two fields are programmed for this type of button: • *Selection:* Select the primary phone that will be associated with this secondary extension button. The primary phone must be on the same node as the phone being programmed. • *Ring When __ Calls At Extension:* This field allows the phone to receive a burst of ringing when "n" number of calls are present at a primary phone, and at least one call is ringing or camped on. That is, a setting of 2 causes the phone to ring when the primary phone receives a third call, while two other calls are ringing. The ring burst repeats periodically as long as "n" number of calls are present at a primary phone. This period is determined by the Digital/IP Secondary Extension Key Alerting Tone timer. If this field is set to 0, the secondary phone never receives the burst of ringing.
SPCL Key	No further programming is required.
SPKR Key	This button type is ***not*** required for digital telephones. No further programming is necessary. This button will be used for turning the phone speaker or headset on and off. Digital Telephones come equipped with a non-programmable SPKR button and do not need this button in their keymap.

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Table 70:   DSS/PKM Button Types and Descriptions (continued)

KEY	DESCRIPTION
Station Speed Dial Key	Select the Station Speed Dial location (0–9) that will be assigned.
System Speed Dial Key	Select the System Speed Dial location (000–999) that will be assigned.
Trunk Group Key	Select the trunk group that will be selected when this button is pressed.
Trunk Key	Select the individual trunk that will be selected when this button is pressed.
Undefined Key	No further programming is necessary. This button can be programmed by the user.
Up Key	No further programming is necessary. This button can be used to scroll forward through displays. When the phone is idle, it can be used for adjusting volume, in place of the VOLUME button. This button type is *not* required for digital telephones. For digital telephones you use the VOLUME DOWN button.
VOLUME DN Key	No further programming is necessary. It can be used for adjusting volume.
VOLUME SAVE Key	No further programming is necessary. This button can be used to save the volume setting. Does not apply to 52xx\53xx phones, because volume changes made on 52xx\53xx phones are automatically saved.
VOLUME UP Key	No further programming is necessary. It can be used for adjusting volume.

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PROGRAMMING DSS/PKM PHONE LISTS

You can add or delete phones that are currently assigned to the DSS/PKM Keymap.

To add to the Phone List:

1. Select System – Phone-Related Information – Key Assignments – DSS/PKM – *<DSS/PKM number>* – **Phones**.

2. Right-click the list. An option box appears.

3. Select **Move to Phone List**. A window appears that asks you to select the type of device to include.

4. Select the device types (you can use the SHIFT or CTRL key to select more than one item), and then click **Next**. The items with details appear. To view items in a list only, click **List**.

5. Select the desired devices, and then select **Move Items**. When you have selected all the devices necessary, click **Finish.** Your selection appears in the station list. To highlight a series of items, hold down **SHIFT** while selecting the first and last item you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

To remove a phone from the list:

1. Highlight the list, and then press **DELETE**. You are prompted to confirm that you want to delete the station.

2. Click **Yes**.

PROGRAMMING DSS OR PKM DEVICES FOR PHONES

You can add a DSS or PKM to a phone.

To program a PKM 16 (mini-DSS), you must use the User Programmable Keys on a phone (see **page 429**).

To associate a phone with a PKM or DSS:

1. Select System – Devices and Feature Codes – Phones – *<phone>* – **Attached Device**.

2. Select the Attached Device current Value, and then use the drop-down list to scroll to **PKM** (for 5224/5324/5330/5340) or **DSS/PKM**.

> **NOTE** Although the PKM option is programmable for all 52xx/53xx phones, PKMs are only supported by 5224/5324/5330/5340 phones. DB Programming does not differentiate between model types.

3. Press **ENTER** or click outside the field to save the change.

To programing a DSS/PKM keymap:

1. Select System – Phone-Related Information – Key Assignments – **DSS/PKM**.

2. Right-click on the list, and then select **Add To DSS/PKM List**. A dialog box appears.

3. Select the starting ID and the number of keymaps (IDs) to add.

4. Click **OK** to add the keymaps to the list.

5. Assign a new description to any of the keymaps by selecting the current description and typing the new information (up to 16 characters) in the text box.

6. Press **ENTER** or select another field to save the change.

7. After you have added DSS/PKM keymaps, double-click on a specific keymap to access the Phones and DSS/PKM Keymap fields.

8. Select the DSS/PKM maps from the DSS/PKM box. Note that PKM 50 (previously known as the Model 8450 DSS) is not applicable to the 5224/5324/8528/8568 phone.

9. Double-click or right-click and choose **Explore DSS/PKM Keymap** to view the keymap dialog.

10. Follow the instructions on the *MiVoice Office 250 DB Programming Help* or "Programming DSS/PKM Keymaps" on **page 440** to automatically or manually populate the DSS/PKM keymap.

11. In the Phones folder, add 5224/5324/8528/8568 or other digital telephones to the list.

PHANTOMS & HOT DESK PROFILES

This section contains the following information:

- **Feature Description** below
- Programming Phantoms and Hot Desk Profiles on **page 449**
- Troubleshooting Phantom Devices on **page 451**

FEATURE DESCRIPTION

The following types of devices are available:

- **Phantom Devices** below
- Hot Desk Profiles for Hot Desking on **page 447**

PHANTOM DEVICES

Phantom devices do not consume a physical hardware address and do not count against the system total device count. Phantom devices are created for the user who typically is not in the office and does not need a phone to retrieve calls or messages.

Although phantom devices do not have a hardware address, they are programmed like other phones. Phantoms consume the same software resources as a regular phone or IP phone, with the exception of the voice paths. Therefore, they can have an impact on system performance. There can be up to 250 phantom devices.

Phantom devices are fully functional, virtual devices on the system. Phantom devices can function with Unified Communicator (UC) to perform advanced call routing tasks without the need for a real desk phone. They can also have a true status, such as idle, Do-Not-Disturb (DND), ringing, and so forth. This allows them to be placed in hunt groups and actually ring.

The following scenarios describe phantom configuration uses:

- A phantom can be configured as a stand-alone device. An employee could have a cell phone to use when they are not in the office but still need to be on the office switch. The administrator could program a phantom for them and the employee (or administrator) could forward the phantom to the cell phone, using UC. That way, callers could still call an extension on the office switch and be able to get the employee on his or her cell phone. In the extreme case, all devices on the switch could be phantoms for users with cell or other phones that are not physically connected to the switch.

- A phantom can be configured as a monitoring device. For instance, a user may put a phantom device in a hunt group and program a phone button to map to that phantom extension. Therefore, a user can monitor when the phantom is ringing, etc., and even answer the call either via a reverse transfer or a secondary extension key.

- A phantom can be configured as a call-park extension. The user may want to set up one or two (or more) phantoms to be extensions at which a call can be parked.

- A phantom can be configured as a general mailbox because phantoms can have associated mailboxes. When a caller calls the phantom, if the phantom is set up to go to voice mail,

the caller can leave a message in this "general" mailbox. However, this configuration is not much different from an unassociated mailbox, with the exception that a phantom mailbox can be accessed off-node (whereas an unassociated mailbox cannot). Note that unassociated mailboxes can be accessed off-node, if a caller calls the appropriate voice mail application and enters the unassociated mailbox extension.

HOT DESK PROFILES FOR HOT DESKING

The Hot Desking feature allows users to share phones or temporarily move to other phones and yet maintain their personal identity and preferred phone configuration. This makes it ideal for businesses with employees who spend only part of their time in the office or who often travel from office to office.

Hot Desking allows a pool of shared phones to be made available to employees instead of requiring that each employee be assigned a dedicated phone. For example, many call centers utilize a "bullpen" setup wherein there are multiple work areas (workstations, cubicles, or offices) each equipped with a phone, but it does not necessarily matter which work area a particular employee uses. A bullpen setup may also have multiple work shifts in which employees use the same phones each time, but on different shifts. In each of these cases, users want to maintain all of their personalized phone settings and have a consistent interface every time they log on regardless of which phone they use.

Similarly, "nomad" users, who travel from location to location or office to office on a regular basis, may use a temporary office and phone at each location — one that is shared by other nomad users. Nomad users also want to maintain all of their personalized phone settings and want each phone they use to behave as if they owned that phone. (Note that in this initial release, Hot Desking does not work across multiple nodes.)

With a Hot Desking solution, a Hot Desk Profile user logs on to a Hot Desk-enabled base phone, and the system then applies the user's phone profile – phone settings such as extension (intercom directory) number, class-of-service, etc., and display preferences such as language, programmable buttons, etc. – to the base phone. See the following section for details.

To use Hot Desking, phones must be designated as Hot Desk phones, and users must have a "Hot Desk Profile." See Programming Phantoms and Hot Desk Profiles on for details.

User Profiles

When a Hot Desk Profile user logs on to a Hot Desk-enabled base phone, the user's phone profile (certain static and dynamic information) moves with the Hot Desk Profile user and is applied to the destination (base) phone.

The following information moves with users when they log on to another phone:

- Associated Extensions
- Call Log Settings
- Class-of-Service Information
- DND Information
- Station Flag Settings
- Forward Information

- Forwarding Paths

- Mailboxes

- Programmable Buttons

- Record-A-Call Capability

- Station Speed-Dial Numbers

- Call Logs (also available via the Call History application on 5320/30/40/60 IP phones)

- Station and Voice Mail Messages

- Message Waiting Indications

- People (contacts) application (available only on 5340 and 5360 IP phones)

- ACD Hunt Group Logins

- Dynamic Extension Express (for main extension)

Note that outgoing extensions and emergency extensions do not move with Hot Desk Profile users; they stay with the Hot Desk-enabled base phone.

Licensing

Hot Desking requires a Hot Desking software license. (Note that this is a single system-wide license and not one per user.)

Multi-Node Support

Hot Desking does not work across multiple nodes.

Procedure

Hot Desk Profile users can log on to any Hot Desk-enabled base phone using the Hot Desk feature code (default is 348), their assigned Hot Desk Profile extension number, and their assigned Hot Desk Profile passcode (i.e., the station passcode for their Hot Desk Profile extension number).

For specific instructions on using the Hot Desk feature, refer to the specific phone user guide.

Restrictions/Exceptions

The Hot Desking feature works with all supported IP phones and digital telephones. It does not, however, work with single-line phones or SIP phones.

Users cannot log on to a phone that has an "active" call in progress, which includes being on hold or having a call on hold. All other call types (including camped on calls and ringing calls) will be dropped if a user logs on to or logs off of a phone. Queue call back requests are maintained, but reminder messages are removed during a Hot Desk logon or logoff.

Any calls placed to the original extension number of a phone that has a current Hot Desk session in progress (i.e., calls to the Hot Desk base phone) will still ring into that phone. Note that these calls will first follow any DND or manual/system forwarding settings that previously existed on that base phone prior to the Hot Desk logon. On the other hand, if a Hot Desk base phone has

Dynamic Extension Express enabled for some reason, then calls to the Hot Desk Base will not ring into the base phone, but they will ring at any associated destinations.

If a user forgets to log off and then attempt to log on elsewhere, the system automatically logs the user off of the original phone (if there are no active calls in progress) and allows the user to log on to the new phone.

NOTE	A Hot Desk logoff does not log users out of any ACD hunt groups they are currently logged into. Unless users want their ACD Hunt Group login to follow them to another phone, they should be instructed to log out of any ACD hunt groups prior to logging off from their Hot Desk session.

The Swap Extensions feature (a phone administrator feature for swapping the extensions of two like devices) cannot be used during a Hot Desk session.

A database save/restore operation does not retain active Hot Desk sessions. Any database restore forces active Hot Desk users to be logged off.

Recommendations

Phone button assignments can present some special challenges with Hot Desking. The combination of potentially different device types, key maps, and user-programmable buttons creates countless permutations. To get the most benefit from button assignments when using Hot Desking, Mitel recommends the following:

- The phone type used to create the user's Hot Desk Profile and the phone type of the Hot Desk base phone the user will normally use for Hot Desking should be the same model to facilitate the button assignments moving to the new device.

- The phone type used to create the user's Hot Desk Profile and the Hot Desk base phone should have the same keymap.

Hot Desk-enabled base phones should be self-labeling phones.

PROGRAMMING PHANTOMS AND HOT DESK PROFILES

For details about programming phantoms and Hot Desk profiles, refer to the *MiVoice Office 250 DB Programming Help*.

NOTE	To support Hot Desking, you must have a Hot Desking software license uploaded to the system.

To use Hot Desking, phones must be designated as Hot Desk phones, and users must have a "Hot Desk Profile." Such profiles can be created from existing phones or phantoms on the system, or they can be created new.

To program a Hot Desk-enabled phone:

1. Select System – Devices and Feature Codes – Phones.

2. Click the phone that supports Hot Desking, and then select **Flags**. The supported phone types are:
   - Digital Telephones
   - 86xx IP Phones

- 52xx/53xx IP Phones
- 8602 Softphones

3. Change the Hot Desking Allowed (to this device) flag to **Yes** for the phone.

Mitel recommends that you also program the Hot Desk feature code (348) under an available LED-equipped programmable feature button on Hot Desk-enabled phones so that users have LED indication when the feature is activated. See the appropriate phone user guide for details on programming feature buttons.

To create a phantom or Hot Desk user:

1. Select System – Devices and Feature Codes – **Phantoms & Hot Desk Profiles**.

2. Right-click in a blank area of the right pane, and then select **Create Phantom** or **Create Hot Desk Profile**. The Create Phantom/Hot Desk Profile dialog box appears.

3. Enter the desired extension number or scroll to the desired number and also enter the number of extensions.

4. *(Optional)* To create a User, select the **Create User(s)** check box. See "Users" on **page 785** for details.

5. Click **OK** to continue. This is the same method used to create other off-node device types.

To program individual phantoms and Hot Desk profiles:

1. Select System – Devices and Feature Codes – **Phantoms & Hot Desk Profiles**.

2. Double-click a phantom or Hot Desk profile to program the following associated options. These options are identical to the ones for phones. See **page 391** for details or the DB Programming Help for details.

- Associated Extensions
- Call Logging
- Class of Service
- Flags
- Forwarding Paths
- Mailboxes
- Programmable Keys
- Record-A-Call
- Account Code Type
- Validated Flag
- Language
- Secondary Language
- Passcode
- Calling Party Name
- Calling Party Number
- Audio for Calls Camped onto this Device

- Audio for Calls Holding for this Device

- Audio for Calls Ringing this Device

- **Associated Hot Desk Device**: Shows which Hot Desk-enabled phone is associated with this Hot Desk Profile. The DB Programming session must be in online mode to view this field so that you can verify if the phone is currently logged on by a Hot Desk user.

TROUBLESHOOTING PHANTOM DEVICES

Table 71 summarizes the troubleshooting strategies recommended for resolving discrepancies with phantom devices.

Table 71:   Phantom Devices Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Phone displays INVALID FEATURE CODE and DND does not disable.	The user accessed remote programming to take a phantom out of DND.	As designed, a phantom device cannot be taken out of DND.
Phone displays INVALID EXTENSION NUMBER and swap phone fails.	A phone administrator accessed the swap phone operation for a phantom device.	As designed, a phantom device cannot swap phones because it does not have a physical hardware address.
When a phone administrator is scrolling through a list of phones, the phone displays CIRCUIT NONE.	An administrator is accessing administrator features for a phantom device.	As designed, there is no device number for a phantom device because it does not have a hardware address.
A phantom cannot forward to the public network.	A phantom does not have access to forward to the public network.	Enable the phantom's Manual Forward to Public Network flag.
	The phantom is not in the CO trunk group's outgoing access list.	Assign the phantom's outgoing extension to the corresponding outside access CO trunk group.
A phantom cannot forward off-node.	The phantom is not in the node trunk group's outgoing access list or node IP connection group.	Assign the phantom's outgoing extension to the corresponding outside access node trunk group or IP connection group.
	The local node does not recognize the off-node device.	Create the off-node device on the local node.
A phantom hunt group member cannot forward to the public network.	Like other forwarded phones, hunt group members cannot forward to the public network.	Phantom hunt group members cannot forward to the public network.

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Table 71:   Phantom Devices Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
When adding more phantoms, the system slows down.	Adding too many phantoms in the current database configuration. Phantoms, though they do not use physical hardware, still use similar processor resources as a physical phone.	Remove some phantoms from the system.
MiVoice Office 250 v2.1 or earlier nodes cannot call a MiVoice Office 250 v2.2 phantom.	The off-node device for the phantom does not exist on the earlier node.	On the earlier node, create an off-node devise for the MiVoice Office 250 v2.2 phantom extension. This will allow the older node to recognize the phantom extension and call it.
The user cannot add more phantoms.	The default phantom value is too low for the current system.	If additional phantoms are necessary, Mitel recommends increasing the phantoms value in On-Line Monitor (OLM) mode so the combined TDM/IP devices and the phantoms value does not exceed the system's license capacity. There are three ways to tell how many devices you have in your system:
		1. Dump the System Device Information to Message Print via an administrator phone.
		2. Use Diagnostics Monitor/System Monitor (v3.2.1 or later) to dump the System Device Information.
		3. Use Mitel DB Programming to view the System Device Information (View/System Device Information).
		If the system needs more phantoms that the system's licensable capacity, contact Mitel Technical Support.
		The system may run slower by increasing the default phantom value.
		Note: Do not use OLM mode unless you are instructed to do so by Mitel support personnel.

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ACCOUNT CODES

This section contains the following information:

- **Feature Description** below

- Programming Account Codes on **page 455**

FEATURE DESCRIPTION

Account codes are 3- to 12-digit codes that can be used in conjunction with the Station Message Detail Recording (SMDR) feature to aid record keeping. Account codes can be assigned to measure telephone use and/or to identify calls for customer billing.

Types of account codes include Standard, Forced, and Optional. The account code, when used, is recorded in the SMDR report as soon as the call is completed. If more than one account code is entered during a call, the last account code that was entered is recorded.

For feature usage instructions, refer to the applicable phone user guide.

DATABASE-PROGRAMMED ACCOUNT CODES

Any phone can be assigned a standard account code or one of six types of forced account codes (four are ARS-dependent). Or, if desired, the phone can have no associated account code. The database can hold up to 256 standard and 256 forced account codes.

OPTIONAL ACCOUNT CODES

Optional account codes can be entered at any time during a call. These user-defined codes are not pre-programmed, but must be within the maximum length set in programming. If entered, optional account codes are printed in the SMDR report for that call in place of standard or forced account codes that may have been used.

ACCOUNT CODES

You can use account codes to force system users to enter a preprogrammed code when placing calls of a certain type. The default database does not contain any account codes, but you add up to 512 account codes. In a networked system, the system validates account codes against the account code table on the user's node. The account code follows the call as it moves from node to node and appears on every SMDR record associated with the call. You cannot use account codes for phantom devices (see **page 417**).

You can assign each phone a standard or forced account code, of which four (two in Europe) are ARS dependent. Or, if desired, the phone can have no associated account code. If you assign a standard account code, you must also designate a specific code (001–512). **Table 72** shows account code types.

Table 72: Account Codes Available for U.S. and Europe

ACCOUNT CODE	DESCRIPTION	U.S.	EUR.
None	Does not require the user to enter an account code. The user may still use optional account codes or enable the Account Code For All Calls Following feature, as desired.	✓	✓
Standard	Automatically appear in the SMDR printout to identify calls from this phone. The phone user is not required to enter any digits during the call.	✓	✓
Forced – All Calls Validated	Requires the user to enter a forced account code before placing an outside call. If the code matches one of the forced account codes in the database, the call is allowed. If the code does not match, the call is blocked.	✓	✓
Forced – All Calls Non-Validated	Requires the user to enter an account code before placing any outside call. The code is not checked against any lists, and the call is allowed as soon as the code is entered.	✓	✓
Forced – Local Toll Calls Validated	Requires the user to enter an account code after dialing a local toll call number when using ARS. If the code matches one of the forced account codes in the database, the call is allowed. If the code does not match, the call is blocked.	✓	
Forced – Local Toll Calls Non-Validated	Requires the user to enter an account code after dialing a local toll call or long distance number when ARS is used. The code is not checked against any lists and the call is allowed as soon as the code is entered.	✓	
Forced – Long-Distance Toll Calls Validated	Requires the user to enter an account code if the system detects that a long distance call has been dialed when ARS is used. If the code matches one of the forced account codes in the database, the call is allowed. If the code does not match, the call is blocked.	✓	
Forced – Long-Distance Toll Calls Non-Validated	Requires the user to enter an account code if the system detects that a long distance call has been dialed when ARS is used. The code is not checked against any lists and the call is allowed as soon as the code is entered.	✓	
Forced – Toll (National) Calls Validated	Requires the user to enter an account code after dialing a national call number when using ARS. If the code matches one of the forced account codes in the database, the call is allowed. If the code does not match, the call is blocked.		✓
Forced – Toll (National) Calls Non-Validated	Requires the user to enter an account code after dialing a national call when using ARS. The code is not checked against any lists, and the call is allowed as soon as the code is entered.		✓

ENABLING THE ACCOUNT CODE FOR ALL CALLS FOLLOWING FEATURE

You can use the Account Code For All Calls Following feature to enter an account code once and apply it to all calls placed from that phone until the feature is canceled. The account code is stored in system memory and is used for all calls made by that phone. It affects other account code programming as follows:

- **Forced:** The "all calls following" account code is used for all calls and the phone user will not be prompted to enter an account code until the "all calls following" code is canceled.

NOTE	If account codes are validated, and the Account Code For All Calls Following is an invalid code, calls will not be allowed at the phone until the code is removed or repro-grammed.

- **Standard:** The "all calls following" account code overrides the standard code.

- **None:** The "all calls following" code will be used as an optional account code.

- **Optional:** In the event that this feature is in effect and an optional account code is also entered, the optional account code will override the "all calls following" account code for that call only. All subsequent calls will be associated with the "all calls following" account code.

To cancel this feature, the feature code is entered without an account code (just press **#** to terminate programming).

PROGRAMMING ACCOUNT CODES

This section provides instructions for programming account codes:

- **Viewing Account Codes** below

- **Programming Forced Account Code Options** below

- **Adding Devices to an Account Code List** below

- Deleting Devices from Account Code Lists on **page 456**

- Assigning an Account Code Type to an Individual Phone on **page 456**

- Setting the Forced Account Code Validated Flag on **page 456**

VIEWING ACCOUNT CODES

To view account codes:

Select System – Phone-Related Information – **Account Codes**.

PROGRAMMING FORCED ACCOUNT CODE OPTIONS

To program forced account code options:

1. Select System – Phone-Related Information – Account Codes – **Forced**.

2. Select one of the following options:

   - **Non-Validated or Validated**: Select either of these options to view a list of account code types. Select an account code type to view a list of phones that have been assigned to that account code list.

   - **Account Codes 001–512**: To enter a new account code, select the current value for the code, and then enter the new digits in the box. The maximum account code length is 12 digits. Account codes can be of varying lengths of the same digits as long as they do not exceed the maximum. For example, if the maximum length is four digits, 1, 11, 111, and 1111 can be entered as separate account codes. Click out of the field or press **ENTER** to save your change.

ADDING DEVICES TO AN ACCOUNT CODE LIST

To add a device to an account code list:

1. Select System – Phone-Related Information – **Account Codes**.

2. Right-click anywhere in the right pane of the window. An option box appears.

3. Select **Add To List**. A window appears prompting for the device type to include.

4. Select the device types (you can use the SHIFT or CTRL key to select more than one item), and then click **Next**. The items with details appear. To view items in a list only, click **List**.

5. Select the appropriate items, then select **Add Items**. When you have added all the devices, click **Finish**. The selections appear in the list. To view programming options, double-click the extension number.

DELETING DEVICES FROM ACCOUNT CODE LISTS

To delete a device from a list:

1. Select System – Phone-Related Information – **Account Codes**.

2. Select the device.

3. Right-click, and then select **Remove Selected Items**.

ASSIGNING AN ACCOUNT CODE TYPE TO AN INDIVIDUAL PHONE

You can assign an account code type to an individual phone:

To assign an Account Code Type to an individual phone:

1. Select – System – Devices and Feature Codes – Phones – (**Local**).

2. Select the extension number.

3. Select **Account Code**, and then select the option (None, Standard, Forced).

4. Click out of the field or press **ENTER** to save your change. Each phone can be assigned *either* a forced or standard account code—or no account code. If you select a phone for standard or forced account code that was previously programmed to require an account code, the previous account code type is replaced by the new code type.

SETTING THE FORCED ACCOUNT CODE VALIDATED FLAG

If you assign a forced Account Code Type to a phone (see the previous section), you can set the Validated flag to check the code entered (flag turned on) or accept any code (flag disabled).

To set the Validated flag:

1. Select – System – Devices and Feature Codes – Phones – (**Local**).

2. Select the extension number.

3. Select **Validated Flag**.

4. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

5. Click out of the field or press **ENTER** to save your change.

ASSISTANTS

The following assistants are available:

- **Configuration Assistant** below

- Conference Assistant on **page 462**

- OfficeLink Assistant on **page 462**

CONFIGURATION ASSISTANT

This section contains the following information:

- **Feature Description** below

- Programming Configuration Assistant on **page 461**

The v5.0 software release added a number of enhancements to the existing Configuration Assistant application, including:

- an option for managing Meet-Me Conferencing access codes (see **page 458**)

- miscellaneous improvements to the existing user interface (see **page 459**)

FEATURE DESCRIPTION

Configuration Assistant is a voice guided configuration portal that provides easy-to-use, remote access to the following end-user phone configuration options:

- Dynamic Extension Express (DEE)

- Do-Not-Disturb (DND)

- Call Forwarding

- Meet-Me Conferencing

- Administrator Functions

Besides providing remote access to these features, Configuration Assistant can also enhance the programming of two-line display phones and single-line phones, as the small display or lack of display makes it more difficult to program these features as originally designed.

| NOTE | Although the original Remote Programming feature is still available, Configuration Assistant is easier to access and use. |

When enabled as a device in DB Programming, the assigned Configuration Assistant extension can be dialed from any phone in the system or accessed remotely using direct transfer, automated attendant, voice mail, and so forth.

Configuration Assistant, which uses system VoIP resources, allows up to five concurrent sessions. If the maximum number of sessions is reached, callers will camp on until a new session is available.

The v4.0 or later software installation process installs "Call Processing" prompts in addition to voice mail Prompts, as shown in the figure below. These CP prompts are used by the Configuration Assistant feature.

Description	Version	Name	Status
Mitel 5000 Basic Rate Interface	4.0.47	cs-brs	Installed
Mitel 5000 Core Utilities	4.0.47	cs-core-utils	Installed
Mitel 5000 Canadian French Call Processing Prompts	4.0.47	cs-cp-cf-pmts	Installed
Mitel 5000 Common Call Processing Prompts	4.0.47	cs-cp-cm-pmts	Installed

Using Configuration Assistant

When a call is placed to Configuration Assistant, the caller is prompted for a phone extension number and passcode (that is, the Remote Programming passcode assigned to the extension). If the caller enters an incorrect extension or times out three times, Configuration Assistant plays a "goodbye" prompt and hangs up. The same holds true if the caller enters an incorrect passcode or times out three times. (The caller is given five seconds to enter an extension number or a passcode before timing out.)

Currently, only phones on the same node as the called Configuration Assistant can be programmed via Configuration Assistant. If a caller tries to program an off-node phone (from the Configuration Assistant's point of view), Configuration Assistant prompts the caller to enter a new extension number. Configuration Assistant is, however, accessible from a remote node via an off-node device.

After the phone extension number and passcode are authenticated, the caller is presented with the following options (see also **page 460** for a summary of the available options):

*   **Dynamic Extension Express (DEE) Option:** This option, which is presented only if the authenticated extension is identified as a DEE user, allows the caller to change their DEE status (on or off) or their mobile phone number. When this option is selected, Configuration Assistant first states the DEE user's current mobile number (as originally programmed in the Users folder). Then, if the user enters a new mobile number, Configuration Assistant now repeats the number that was entered and prompts the user to accept the new number or re-enter another number.
    *   Enable (only if DEE is disabled)
    *   Disable (only if DEE is enabled)
    *   Change mobile phone number

*   **DND Status Option:** This option allows the caller to change the DND status (on or off) of the authenticated extension number. (There is no option to provide a specific DND status message.) When this option is selected, Configuration Assistant first states whether DND is enabled or not (for example, "Do-Not-Disturb is disabled") and then offers the following DND functions:
    *   Enable (only if DND is disabled)
    *   Disable (only if DND is enabled)

*   **Call Forwarding Option:** This option allows the caller to change the manual call forwarding status (on or off) of the authenticated extension number. (There is no option to provide a

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specific call forwarding condition, such as no answer or busy.) When this option is selected, Configuration Assistant first states current call forwarding status. Configuration Assistant then indicates one of the following: call forwarding is disabled, calls are forwarded to an outside number, calls are forwarded to an internal extension, or calls are forwarded to voice mail. If the user enters a new call forwarding destination, Configuration Assistant repeats the number and prompts the user to accept the new number or re-enter another number. Configuration Assistant then states whether call forwarding is enabled or not (for example, "Call forwarding is enabled").

- Disable (only if forwarding is enabled)
- Forward calls to an internal extension number
- Forward calls to an outside phone number
- Forward calls to voice mail

- **Meet-Me Conferencing:** This option, which is presented only if the system is configured for Meet-Me Conferencing, allows callers (with a programmed main extension in the Users folder) to manage their Meet-Me Conferencing access codes. When this option is selected, Configuration Assistant offers the following Meet-Me Conferencing options:
  - Create a new access code
  - Delete an existing access code
  - List existing access codes
  - Send an e-mail message with the access code list (only if the system E-Mail Gateway settings are configured (see "E-Mail Gateway Programming Options" on **page 942** for details), and only if an e-mail address is configured under the Users folder in DB Programming; see "E-mail Address" on **page 788** for additional information on the E-mail Address field).

- **Administrator Option:** This option, which is presented only if the authenticated extension is identified as a designated administrator, allows the caller to change the system's night ring status. (There are currently no other administrator functions available in this release.) When this option is selected, Configuration first states whether night ring is enabled or not (for example, "Night ring is enabled") and then offers the following administrator functions:
  - Enable night ring (only if night ring is currently disabled)
  - Disable night ring (only if night ring is currently enabled)

The figure below summarizes the options currently available using Configuration Assistant.

Configuration Assistant				
Dynamic Ext. Exp.	DND	Forwarding	Meet-Me Conf.	Admin
Enable Disable Change Mobile	Enable Disable	Disable Forward to Ext. Forward to No. Forward to VM	Create Acc. Code Delete Acc. Code List Access Codes Send E-Mail	Enable Night Ring Disable Night Ring
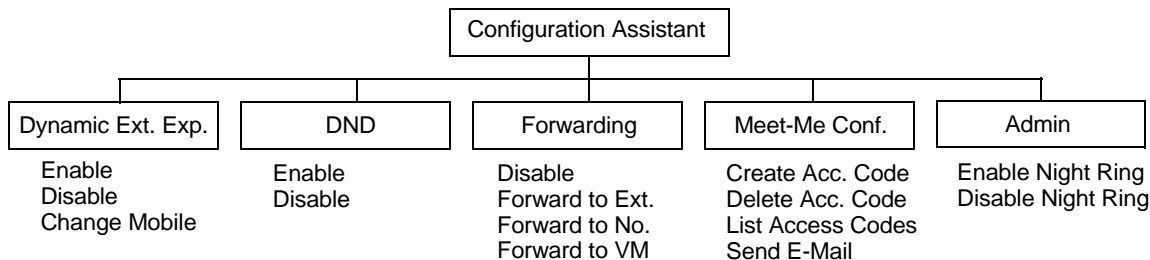
For feature usage instructions, refer to the applicable phone user guide.

PROGRAMMING CONFIGURATION ASSISTANT

This section describes how to program Configuration Assistant.

To create a Configuration Assistant extension:

1. Select System – Devices and Feature Codes – **Assistants**.

2. Right-click and select **Create Configuration Assistant**. The Create Configuration Assistant dialog box appears.

3. Select or type the extension in the Starting Extension drop-down list box. You can select one extension. On a multi-node system with more than one Configuration Assistant, the Configuration Assistants should have unique extensions so they can be accessed from other nodes.

4. Click **OK**. The extension appears in the Assistants folder as shown below. The Enable Configuration Assistant indicates whether the Configuration Assistant feature is enabled. It is set to Yes by default.
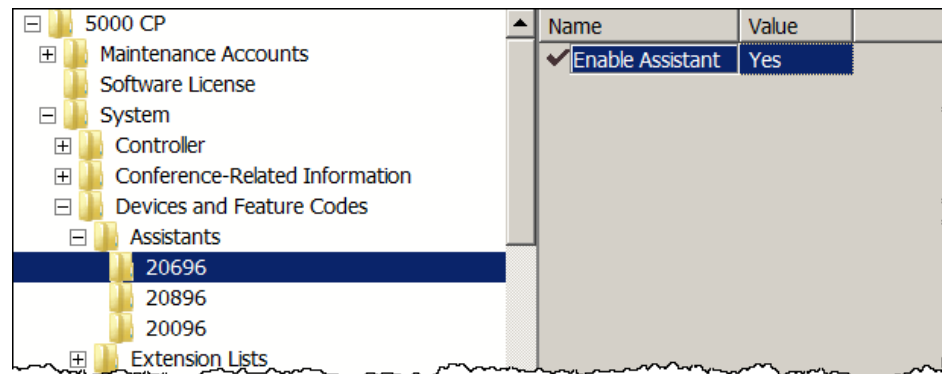
To disable Configuration Assistant:

1. Select System – Devices and Feature Codes – Assistants – <***extension***>.

2. Change the Enable Configuration Assistant flag to **No**.

Export/Import Devices

The Configuration Assistant device type is treated as a phantom for import/export and is imported/exported as an off-node phantom. Therefore, this device type is not included in the list of Device Types in the Export/Import Devices dialog box under Operations.

Selection List

Configuration Assistant devices are included in the selection lists in the same places as the Auto Attendant. The figure below is an example of the Hunt Group Members List.
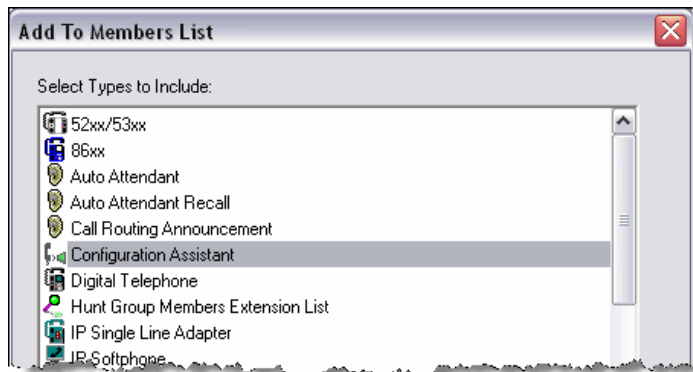
CONFERENCE ASSISTANT

The Conference Assistant is a device with a dialable extension that users will call to access Meet-Me Conferences. The default extension uses the Hunt Group base. The default description and username are "Conference Assistant" and "CONFERENCE," respectively. Only one Conference Assistant may be programmed per system and there is no Conference Assistant programmed by default. The Conference Assistant can be exported/imported to/from other nodes as a Phantom device. If the system does not have a Meet-Me Conferencing software license, the Configuration Assistant appears with a red "X." See "Meet-Me Conferencing" on **page 283** for details.

To create a Conference Assistant extension:

1. Select System – Devices and Feature Codes – **Assistants**.

2. Anywhere in the right pane, select **Create Conference Assistant**. The Create Assistant dialog box appears.

3. Select or type the extension in the Starting Extension drop-down list box. You can select one extension. On a multi-node system with more than one Assistant, the Assistants should have unique extensions so they can be accessed from other nodes.

4. Click **OK**. The extension appears in the Assistants folder.

OFFICELINK ASSISTANT

The OfficeLink Assistant supports the OfficeLink feature for Mitel Unified Communicator Advanced (UCA) v4.0 (now known as MiCollab Client). The MiCollab Client OfficeLink feature allows users to place calls, using MiCollab Client, from one of their ring group devices. Ring group devices include the devices configured for the user on the communications platform. Adding an OfficeLink Assistant on the MiVoice Office 250 allows MiCollab Client users to place OfficeLink calls from any of their ring group devices (including Mobile). If the OfficeLink Assistant is not present (or if using software older than 5000 CP v5.0), then MiCollab Client users can only place OfficeLink calls from their desk phone and MiCollab Client Softphone devices.

The OfficeLink feature is available from the MiCollab Client Web Portal, Mobile Portal, and Mobile for BlackBerry client interfaces. When users activate the OfficeLink feature, they must specify the following:

- The number to call
- The device to place the call from

The response from the MiVoice Office 250 varies based on the device the user has selected to place the call:

- **Desk Phone and MiCollab Client Softphone**: The system immediately places a call to the specified number from the specified device. This behavior is also known as click to call.

- **External Device (Home, Mobile)**: The system places a call to the device the user selected. After the user answers the call on the device, the system immediately places a call to the specified number. This behavior is also known as remote click to call.

For more information about the MiCollab Client OfficeLink feature, refer to the MiCollab Client Administrator Guide.

To create an OfficeLink Assistant extension:

1. Select System – Devices and Feature Codes – **Assistants**.

2. Anywhere in the right pane, select **Create OfficeLink Assistant**. The Create Assistant dialog box appears.

3. Select or type the extension in the Starting Extension drop-down list box. You can select one extension. On a multi-node system with more than one Assistant, the Assistants should have unique extensions so they can be accessed from other nodes.

4. Click **OK**. The extension appears in the Assistants folder.

PHONE MESSAGES

This section contains the following information:

- **Do-Not-Disturb** below

- Changing Do-Not-Disturb Messages on **page 465**

- Reminder Messages on **page 466**

- Changing Reminder Messages on **page 467**

DO-NOT-DISTURB

Placing a phone in Do-Not-Disturb (DND) halts all pages, incoming intercom calls, camped-on calls, and transferred calls to that phone. Queue callbacks, recalls, and direct ring-in calls are not blocked. Another user calling the phone while it is in Do-Not-Disturb hears a repeating signal of four fast tones and a pause. Display phones show the Do-Not-Disturb message. The caller cannot camp on, but can queue or leave a message at the phone.

For feature usage instructions, refer to the applicable phone user guide.

NOTE	Direct ring-in calls that are forwarded to a phone in DND will ring the DND phone in accordance with the rules of DND.

If desired, individual phones can be prevented from using DND by disabling the feature option in the database.

If a hunt group phone is in DND, calls to the user's hunt group do not cause the phone to ring, but the individual trunk button will flash if all other phones in the hunt group are busy, forwarded, have hunt group remove enabled, or are in DND. Hunt group announcement and overflow stations can use DND to block hunt group calls by using DND.

Phones may be given DND override permission. These phones, when reaching a phone in DND, can enter the Do-Not-Disturb Override feature code (373) to place a nonhandsfree intercom call to the phone. Single line phones cannot be enabled to use the Do-Not-Disturb Override feature.

When a phone is placed in Do-Not-Disturb, the user may select one of 20 system-stored messages that will appear on the top line of the display (unless DND is enabled while the user is on a call or off-hook, in which case message 1 is automatically selected). When a phone in DND is called by a display phone user, the caller sees the selected message.

Do-Not-Disturb messages can be programmed directly in the database or by using an Administrator phone. For more information, refer to the *MiVoice Office 250 Phone Administrator Guide,*.

Default DND messages are shown on **page 466**.

DND ON A NETWORK

DND works as follows in a network setting:

- Each node has its own list of DND messages that can be used only on that node.

- An intercom caller will see another user's DND display when calling across nodes.

- The network allows DND override across nodes.

The system has default DND messages in both the primary and secondary languages. The language of the messages seen by the phone user, both when programming his or her phone and when calling another phone that is in DND, is determined by the phone's programmed language. That is, if a phone is programmed for Japanese, the user sees only Japanese messages when programing a Do-Not-Disturb message. If that phone calls an American English-programmed phone that has selected DND message 02, the Japanese phone's user sees the Japanese version of message 02. You must use an administrator phone to reprogram messages that use Japanese characters. Refer to the *MiVoice Office 250 Phone Administrator Guide,* for more information.

NOTE	Only the MiVoice Office 250 8660 IP phone can be used for viewing Japanese prompts. Other phones do not support Japanese displays. For complete information about multilingual capability, see "Multilingual Capability" on **page 314**.

DND OVERRIDE

If a phone is enabled for Do-Not-Disturb override, the following procedure can be used to break through Do-Not-Disturb and complete the call.

To use Do-Not-Disturb override (phones with this feature enabled only):

When you hear Do-Not-Disturb tones while placing an intercom call, press the Special button, followed by the Do-Not-Disturb Override feature code (373). If the called phone is idle, the call rings as a private intercom call. If the called phone is busy, your call camps on.

CHANGING DO-NOT-DISTURB MESSAGES

Phones show DND messages in English, British English, Canadian French, Katakana (Japanese), and Spanish characters. For complete information about DND, see "Do-Not-Disturb" on **page 292**.

NOTES	The Japanese language is not supported on 52xx/53xx-series phones.
	You must use an administrator phone to program Japanese messages. For more information, refer to the *MiVoice Office 250 Phone Administrator Guide*, part number 580.8001.
	For complete information about multilingual capability, see "Multilingual Capability" on **page 314**.

When changing DND messages, you should keep the meanings for the messages in all languages the same. This allows phone users to select the message in the same location for either primary or secondary languages. For example, if you change the DND message "02" to "PAGE ME" in the English language, you should program a similar message for message "02" in the other languages.

The default Do-Not-Disturb messages are shown in **Table 73**.

Table 73:   Default DND Messages

CODE	DEFAULT MESSAGE	CODE	DEFAULT MESSAGE
01	Do-Not-Disturb	11	Out of Town 'Til
02	Leave a Message	12	Out of Office
03	In Meeting Until	13	Out Until
04	In Meeting	14	With a Client
05	On Vacation/Holiday 'Til	15	With a Guest
06	On Vacation/Holiday	16	Unavailable
07	Call Me At	17	In Conference
08	At the Doctor	18	Away from Desk
09	On a Trip	19	Gone Home
10	On Break	20	Out to Lunch

To change a DND message:

1. Select System – Phone-Related Information – **Messages**.

2. Double-click the language for the DND messages that you want to change.

3. Double-click **DND**.

4. In the **Value** column, type the new message in the text box (up to 16 characters, including spaces).

5. Click out of the field or press **ENTER** to save your change.

REMINDER MESSAGES

Reminder messages are set to signal a phone at a specified time. The user can select one of 20 different messages and set the reminder time up to 24 hours in advance. These messages can be reprogrammed by the system administrator or by using an administrator phone.

NOTE	Reminder messages are available on Mitel proprietary phones only.

At the programmed time, the reminder message signals the phone with eight short tones. A display phone shows the message until it is canceled. A non-display phone receives tones only. If the phone is busy, the user still hears the tones and the message displays for 10 seconds during the call, then the display returns after the user hangs up. Reminder displays interrupt, but do not affect, programming.

For feature usage instructions, refer to the applicable phone user guide.

Messages can be changed by the installer or with an administrator station, if desired. They are limited to 16 characters. Default messages are shown in **Table 74**.

Table 74:   Reminder Messages

01 MEETING	11 CALL ENGINEERING
02 STAFF MEETING	12 CALL MARKETING
03 SALES MEETING	13 CALL ACCOUNTING
04 CANCEL MEETING	14 CANCEL DND
05 APPOINTMENT	15 CANCEL CALL FWD
06 PLACE CALL	16 TAKE MEDICATION
07 CALL CLIENT	17 MAKE RESERVATION
08 CALL CUSTOMER	18 REVIEW SCHEDULE
09 CALL HOME	19 LUNCH
10 CALL CORPORATE	20 REMINDER

In a network, each node has its own list of reminder messages that can be used only on that node.

The system has default reminder messages in both the primary and secondary languages. However, messages using Japanese characters can be reprogrammed only through an Administrator's phone. Refer to *MiVoice Office 250 Phone Administrator Guide,*.

The language of the messages seen by the phone user is determined by the phone's programmed language. That is, if a phone is programmed for Japanese, the user will see only Japanese messages when programing a reminder message.

Only the 8660 IP phone can be used for viewing Japanese prompts. Other phones do not support Japanese displays.

CHANGING REMINDER MESSAGES

Phones show Reminder messages in English, British English, Canadian French, Katakana (Japanese), and Spanish characters.

NOTES	The Japanese language is not supported on 52xx/53xx-series phones.
	You must use an administrator phone to program Japanese messages. For more information, refer to the *MiVoice Office 250 Phone Administrator Guide*, part number 580.8001.
	For complete information about multilingual capability, see "Multilingual Capability" on **page 314**.

When changing Reminder messages, you should keep the meanings for the messages in all languages the same. This allows phone users to select the message in the same location for either primary or secondary languages. For example, if you change the Reminder message "02" to "PAGE ME" in the English language, you should program a similar message for message "02" in the other languages.

The default Reminder messages are shown in **Table 75**.

Table 75:   Default Reminder Messages

CODE	DEFAULT MESSAGE	CODE	DEFAULT MESSAGE
01	Meeting	11	Call Engineering
02	Staff Meeting	12	Call Marketing
03	Sales Meeting	13	Call Accounting
04	Cancel Meeting	14	Cancel DND
05	Appointment	15	Cancel Call Fwd
06	Place Call	16	Take Medication
07	Call Client	17	Make Reservation
08	Call Customer	18	Review Schedule
09	Call Home	19	Lunch
10	Call Corporate	20	Reminder

To change a Reminder message:

1. Select System – Phone-Related Information – **Messages**.

2. Double-click the language for the DND messages that you want to change.

3. Double-click **Reminder**.

4. In the **Value** column, type the new message in the text box (up to 16 characters, including spaces).

5. Click out of the field or press **ENTER** to save your change.

SYSTEM FORWARDING PATHS

You can program up to 200 unique forwarding paths (001–200). When System Forwarding is selected from the Phone-Related Information programming area, the list of forwarding paths 1–200 appears. For more information about System Forwarding, "Forwarding – System Forwarding" on **page 298**.

NOTE	After programming, end users must enable and disable this feature from their phones using feature code 354 (System Forwarding on/off). Refer to the applicable phone user guide for more information.

This area is used for assigning forwarding points to the system forwarding paths. You can then program phones to use the forwarding paths in phone programming (see **page 401**).

Each forwarding path can have a distinctive description (of up to 20 characters) and four forwarding points. The forwarding points can be local phones, voice mail ports, or hunt groups, or they can be off-node devices. Program the fields for the forwarding paths as follows:

To program a forwarding path description:

1. Select System – **Phone-Related Information**.

2. Double-click **System Forwarding Paths**.

3. In the applicable **Description** column, type a name for the forwarding path, up to 20 characters, in the box.

4. Click out of the field or press **ENTER** to save your change.

To program a forwarding point:

Use one of the following methods:

  Method A

  a. Select the current value, and then enter the new value in the text box.

  b. Press **ENTER**. A screen appears displaying what is associated with the number entered.

  c. Click **OK**. The new number appears in the field.

  Method B

  a. Right-click the existing Forwarding Point. An option box appears.

  b. Select **Change Forwarding Point**. A window appears prompting for the device type to include.

  c. Select the desired device, and then click **Next**. The devices with details appear. To view devices in a list only, click **List**.

  d. Select the desired device, and then click **Finish**. The selection appears in the appro-priate Forwarding Point field.

SYSTEM SPEED DIAL

This section contains the following information:

- **Feature Description** below

- Programming System Speed Dial Numbers on **page 471**

- Creating a CSV File on **page 472**

- Importing System Speed Dial Entries from CSV Files on **page 473**

- Exporting System Speed Dial Entries on **page 475**

FEATURE DESCRIPTION

Speed Dial allows users to dial stored telephone numbers quickly. Up to 1000, 48-digit System Speed Dial numbers can be stored in system memory. If desired, an identifying name can also be stored with each number. Phones programmed with access to this feature can dial any of the numbers on the list. The Administrator maintains the System Speed Dial list.

See also "Caller ID, DNIS, and ANI" on **page 172** and "Directory of Intercom, Speed Dial, and Feature Codes" on **page 290**.

The MiVoice Office 250 supports up to 1000 System Speed Dial numbers. The default database contains only System Speed Dial bin #000 because this is used by the system to establish links to other Speed Dial bins. You may create new bins, individually or in batches, as needed, through DB Programming.

NOTE	Only 500 System Speed Dial numbers can be created manually at one time. If you need more, create them in two stages. 1000 System Speed Dial numbers can be imported at a time.

Each node in a network has its own System Speed Dial numbers. System Speed Dial numbers can be used only on the node where they are programmed.

To keep System Speed Dial numbers confidential, some or all can be programmed as non-display numbers. Non-display numbers can be used by any user but are displayed only on the programming user's phone. Non-display numbers cannot be redialed or saved as Station Speed Dial numbers at a display phone. Non-display numbers will appear in the SMDR record.

System Speed Dial names can be programmed by an Administrator using specific language characters, or a combination. The programmed language for the phone does not affect the characters that can be viewed. That is, no matter what language the phone uses or which characters are in the name, the user will be able to see the Speed Dial names exactly the way they were programmed. For complete information about multilingual capability, see "Multilingual Capability" on **page 314**.

System Speed Dial numbers are subject to toll restriction unless a system-wide option has been enabled that allows any phone to dial any System Speed Dial number regardless of toll restriction.

The System Speed Dial numbers are stored using location codes (000–999). When dialed, the numbers appear on a display phone unless they have been programmed as non-display

numbers. Display phone users can also view System Speed Dial numbers and names without dialing; however, with non-display numbers, only the name is displayed.

System Speed Dial codes 000–999 can be stored in programmable phone buttons by the system administrator. These button codes allow one-button dialing of System Speed Dial numbers by users.

The System Speed Dial numbers and names are stored in battery-backed RAM and will not be erased in the event of a power failure.

The System Speed Dial numbers and names are programmed by the system administrator or at any Administrator phone. For Administrator programming instructions, refer to the *MiVoice Office 250 Phone Administrator Guide,*.

System Speed Dial bins may also be programmed from an administrator phone, using the System Speed Dial feature code (9801). If the user enters a bin that is currently not equipped, one will be automatically equipped and ready for programming. This is transparent to the user at the phone.

When converting from an older software version to a newer version, the conversion program will automatically unequip all System Speed Dial bins that have an empty name and number, with the exception of bin #000.

A user at a phone cannot program System Speed Dial bins when there is an active DB Programming session, and no one can log into DB Programming when System Speed Dial bins are being programmed from a phone.

System Speed Dial numbers can only be used on the node where they are programmed. Each node in the network must have its own System Speed Dial numbers.

PROGRAMMING SYSTEM SPEED DIAL NUMBERS

This section describes how to program a System Speed Dial number.

NOTE Only 500 System Speed Dial numbers can be created manually at one time. If you need more, create them in two stages. 1000 System Speed Dial numbers can be imported at a time.

To program a System Speed Dial number:

1. Select System – **Phone-Related Information**.

2. Double-click **System Speed Dial**.

3. Complete the following options:

    •   **Name**: If desired, you can program a name for the speed-dial number. Each name can have up to 16 characters.

    To program the name:

    a. Select the current value, and then enter the new name in the text box.

    b. Click out of the field or press **ENTER** to save your change.

- **Number**: Numbers can include up to 48 digits and can include digits (0–9, *, and #), timed pauses, or hookflashes/recalls. Timed pauses and/or hookflashes/recalls are used when entering a series of numbers, such as access codes, security codes, and numbers for specialized common carrier (SCC) dialing (U.S. only). To include a pause in the number, enter the letter P for a pause. To include a hookflash/recall, enter F. The pause length represented by the P is determined by the Pause timer. Each pause or hookflash (recall) is considered one of the 32 digits.

To program the number:

a. Select the current value, and then enter the new digits in the text box.

b. Click out of the field or press **ENTER** to save your change.

- **Private Number**: Determine which System Speed Dial number locations may be viewed on display phones. Non-display numbers do not appear on display phones when dialed and cannot be redialed at display phones. Non-display numbers will appear in the SMDR when dialed. To allow the speed-dial number to be displayed when dialed, enable the flag. Disable the flag to make it a non-display number.

To enable the Private Number flag:

a. Select the current value, and then select the check box. The field changes to **Yes**. To disable the option, clear the check box.

b. Click out of the field or press **ENTER** to save your change.

CREATING A CSV FILE

You can create system speed dial entries in a CSV file and import/export the entries to/from DB Programming. See "Considerations when Creating a CSV File" on for more information about CSV files.
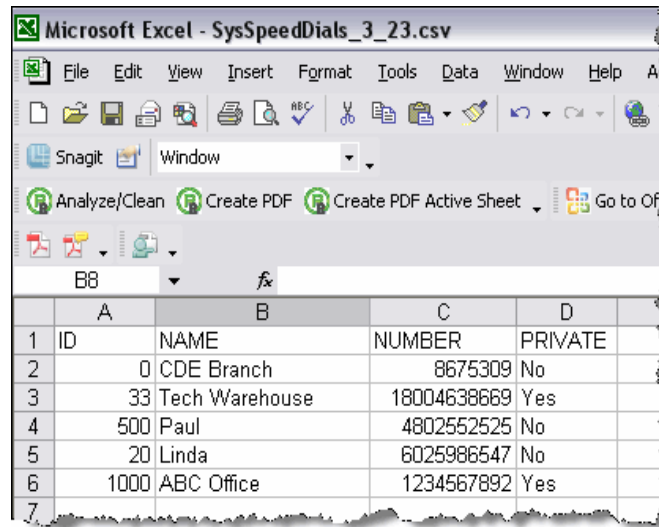
> **NOTE** Up to 1000 System Speed Dial numbers can be imported at a time.

The following instructions describe how to create a system speed dial file.

To create a CSV file:

1. Open the template file, `SpeedDials.csv`, located in C:\Program Files\Mitel\5000\Templates or C:\InterTel\CS5000\Templates (if the system was updated from v3.0 or earlier) on the DB Programming computer.

2. Type the information that you want to import. The fields include:
   - **ID**: ID number for the system speed dial.
   - **NAME**: Description of the speed dial number.
   - **NUMBER**: Number to be dialed when this speed dial is used.
   - **PRIVATE**: Flag to indicate whether or not this speed dial is a private number.
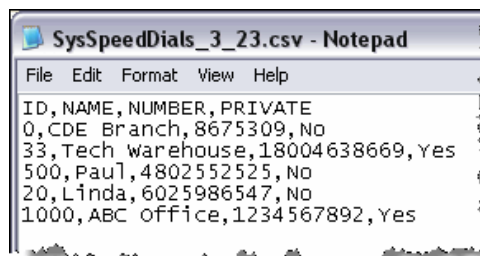
The following is an example of the system speed dial information listed in a Microsoft Excel spreadsheet.

3. Save the data as a CSV file. The following is an example of the file data after it is saved.

IMPORTING SYSTEM SPEED DIAL ENTRIES FROM CSV FILES

You can import system speed dial entries from CSV files. Keep the following considerations in mind when importing a CSV file:

> **NOTE** Up to 1000 System Speed Dial numbers can be imported at a time.

- DB Programming allows duplicate entries (with the exception of the ID). In other words, the import feature allows different IDs with the same Name, Number, and Private Number flag to be created. It is up to you to review the programming and remove duplicates if desired. Duplicate entries, however, have no negative impact other than possible confusion and wasting a System Speed Dial entry.

- When you manually edit system speed dial entries after the CSV file has been imported, do not overwrite the information as the changes may cause conflict.

- DB Programming does not allow you to change a system speed dial ID after it is created. You must be sure that the IDs specified in the CSV file are the IDs that you want to use.

- If any digits are present or any headers are missing in the first line of the file, DB Programming assumes there is no header in the file. The fields are defaulted as specified below:

- **ID**: Defaults to the next available if not provided. If an ID conflicts with an existing speed dial ID, it overwrites the current entry. If the maximum number of entries is reached during the import, the import stops and a message appears indicating that the entries have reached maximum numbers that you can create.

- **NAME**: Defaults to blank if not provided.

- **NUMBER**: Defaults to blank if not provided.

- **PRIVATE**: Defaults to No if not provided.

To import system speed dial entries from a CSV file:

1. Select System – Phone-Related Information – **System Speed Dial**.

2. Right-click in the right pane, and then select **Create System Speed Dials from File**.

3. When the Select the CSV file to import dialog box appears, select the file, and then click **Open**.
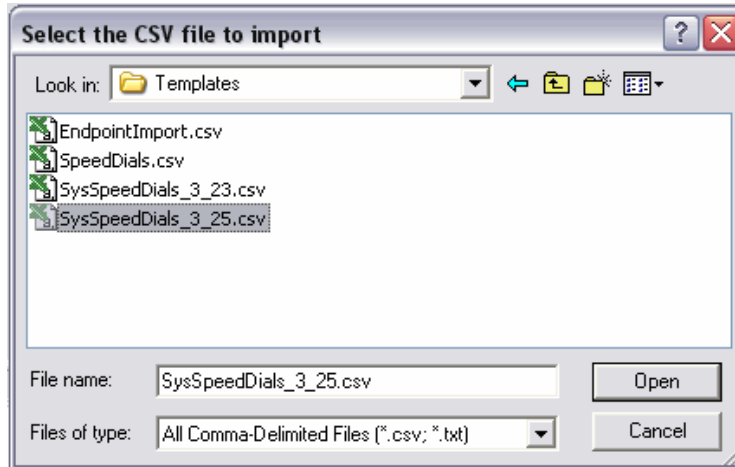
4. When a dialog box appears indicating whether you want to erase all existing System Speed Dials prior to importing, click **Yes** to erase all existing system speed dial numbers prior to performing the import or **No** to verify if there are any duplicates.

5. If there are duplicates, DB Programming asks you how you want to proceed with them. Click **Yes** to overwrite the existing system speed dials that conflict with the import file entries, click **No** to skip the duplicate entries, or click **Cancel** to abort the import.

6. If there were duplicates and you selected **Yes** (overwrite them) or **No** (skip them), a dialog box appears showing you which entries were overwritten or skipped (up to 200 entries).

Below is an example of the skipped entries. By default, the entries are sorted by the ID number.
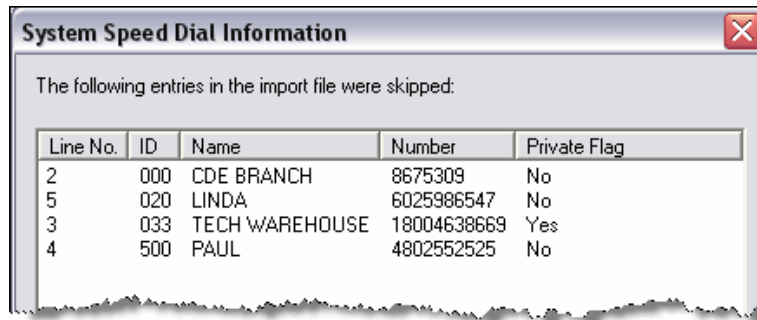
7. Click **OK**.

8. If any entries in the import file are invalid, DB Programming asks you wether you want the system to attempt to fix them during the import. Click **Yes** to fix the entries or **No** to skip the invalid entries, or click **Cancel** to abort the import.
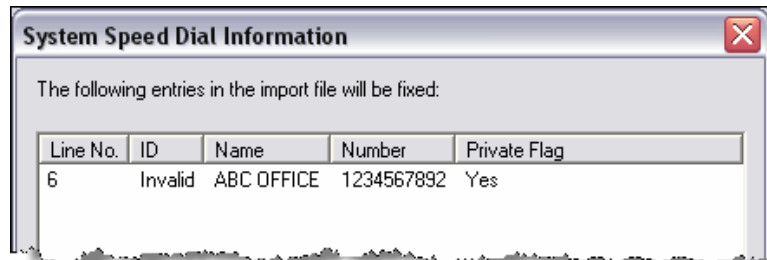
9. Click **OK**.

EXPORTING SYSTEM SPEED DIAL ENTRIES

System speed dial entries may be exported to a CSV file in a similar fashion to importing the system speed dial entries.

To export system speed dials to a CSV file:

1. Select System – Phone-Related Information – **System Speed Dial**.

2. Do one of the following:

To export selected items:

a. Select the items that you want to export.

b. Right-click, and then select **Export Selected Items**.

To export all items:

• Right-click anywhere in the right pane, and then select **Export All Items**.

3. When the Export Speed Dials dialog box appears, rename the file name, and then click **Save**. By default, the file is saved in C:\Program Files\Mitel\5000\Templates on the DB Programming computer.
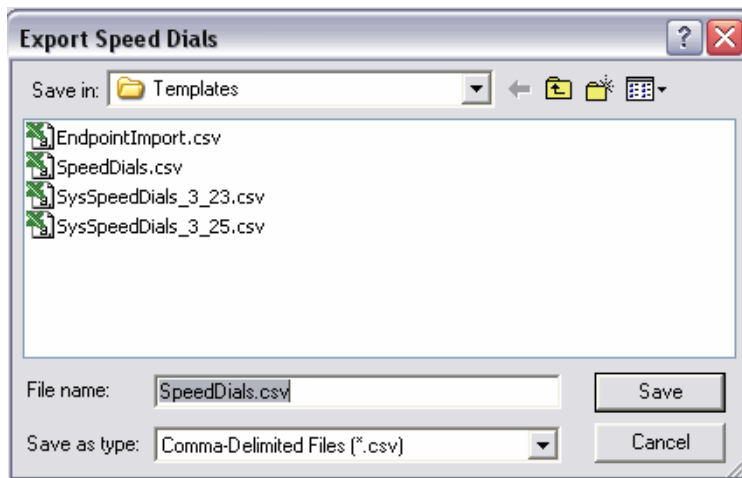
STATION SPEED DIAL

A phone user can program up to 10 Station Speed Dial numbers of 16-digits each. Single line phones use Speed Dial location codes (0–9). Display phones use Speed Dial buttons, if programmed in the phone's keymap, or location codes 0–9.

Together, the lamps in the phone Station Speed Dial buttons create a busy lamp field that indicates the status of the phones programmed under the buttons. For more information about busy lamp fields, refer to the "Specifications" chapter in the *MiVoice Office 250 Installation Manual* . Speed Dial buttons can contain outside telephone numbers, feature codes, phone extension numbers, or hunt group pilot numbers.

Station Speed Dial codes 0–9 also can be stored under user-programmable buttons to create Speed Dial buttons.

If desired, outside telephone numbers can be preceded with a trunk access code to allow one-button dialing of outside telephone numbers. For example, a button programmed with "89619000" would select a trunk using the Outgoing Calls feature code (8) and then dial 961-9000. A phone extension number can be preceded with a pound (#) to always Speed Dial private intercom calls to the phone. Or, a "4" may be entered before a phone extension number or hunt group pilot number to quickly reverse transfer, or pick up, calls from that phone or hunt group. If either of these options is used, normal handsfree intercom calls cannot be placed using that Station Speed Dial location or Speed Dial button, and the Speed Dial button will not show the phone's status.

An outside telephone number can be preceded by a trunk access code for easier trunk selection and number dialing. Phone users can also program pauses and/or hookflashes into the stored outside telephone numbers. For example, the number can contain an SCC local number, a pause, and an access code. When programming Speed Dial numbers, each hookflash and each pause is considered one digit. The durations of the hookflash and the pause are determined by the programmable CO Hookflash and Pause Dialing Digit Length timers.

Display phone users can program a name to be associated with each Station Speed Dial number. Speed Dial names can contain up to 16 characters. To program Speed Dial names, keypad buttons are used to enter the desired letters, numbers, and punctuation. Among the IP phones, only 8660 phones can display Japanese characters. For feature usage instructions, refer to the applicable phone user guide. For complete information about multilingual capability, see "Multilingual Capability" on .

ADMINISTRATOR PHONE DB PROGRAMMING PASSCODE

This passcode is used when programming through an administrator phone. For more information about using the administrator phone, refer to the following guides:

- MiVoice Office 250 Phone Administrator Guide, part number 580.8001

- MiVoice Office 250 Unified Voice Messaging Administrator Guide, part number 580.8009

To program the passcode:

1. Select System – **Phone-Related Information**.

2. Right-click **IP/Digital Telephone Database Programming Passcode**, and then click **Edit Password**. The following dialog box appears.

```
┌─────────────────────────────────────────────────────┐
│ Edit IP/Digital Telephone Database Programming Passc... [X] │
├─────────────────────────────────────────────────────┤
│                                                       │
│   New Passcode:  [          ]      ┌──────────┐       │
│                                    │    OK    │       │
│   Confirm Passcode: [          ]   └──────────┘       │
│                                    ┌──────────┐       │
│                                    │  Cancel  │       │
│                                    └──────────┘       │
└─────────────────────────────────────────────────────┘
```
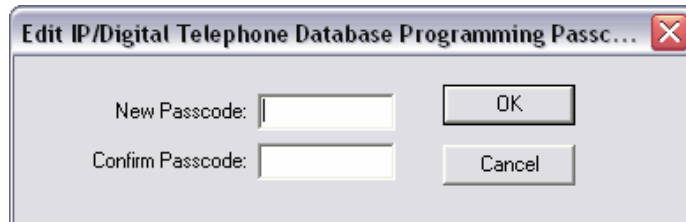
3. In the **New Passcode** box, type the new passcode (up to 8 digits, using digits 0–9). Typed characters appear as asterisks (***).

4. Retype the passcode in the **Confirm passcode** box.

5. Click **OK** to exit and save the passcode. If the entered passcodes match, you return to the Passcode field. If not, you must re-enter the new passcode and verify it again. If you make a mistake while entering the passcode or want to leave it unchanged, click **Cancel**.

NOTE	To provide system security and prevent unauthorized access to the system database, you should enter a passcode for the administrator phone that is difficult to guess. For example, you should not use the phone extension number or several repeated digits. You or the phone user should periodically change the passcode.

MESSAGE CENTERS

A phone can be designated as a message center and assigned a list of phones that it will serve. When you select **Message Centers** from the Phone-Related Programming area, you can view a list of the existing message centers.

If you double-click a specific message center, you can view the list of phones it serves.

To assign the phones served by the message center:

1. Select System – Phone-Related Information – **Message Centers**.

2. Double-click the Message Center extension.

3. Right-click anywhere in the right side of the window. An option box appears.

4. Select **Move To List**. A window appears prompting for the device type to include.

5. Select the device type(s), and then click **Next**. A list appears. You can view them in a list by selecting the List button or view details by selecting the Details button. To select a series of items, hold down **SHIFT** while selecting the first and last items in the range. To select two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

6. Select the appropriate items, then select **Move Items**. When you have added all the devices, click **Finish**. The selections appear in the list. To view programming options, double-click the extension number.

To delete a device from the list:

1. Select the item(s).

2. Right-click and select **Move to NONE List**.

You can also assign message centers in phone programming by selecting the phone Associated Extensions and Message Center options and selecting the phone that will serve as the message center. See **page 392**.

ATTENDANTS

A phone can be designated as an attendant and assigned a list of phones that it will serve. When you select Attendants from the Phone-Related Programming area, you can view a list of the existing attendants.

If you double-click a specific attendant, you can view the list of phones it serves.

To create an attendant:

Enable the phone Attendant flag, as described on **page 419**.

To assign the phones served by the attendant:

1. Select System – Phone-Related Information – **Attendants**.

2. Select the extension number.

3. Right-click anywhere in the right pane. An option box appears.

4. Select **Move To List**. A window appears prompting for the device type include.

5. Select the device type(s), and then click **Next**. A list appears. You can view them in a list by selecting the List button or view details by selecting the Details button. To select a series of items, hold down **SHIFT** while selecting the first and last items in the range. To select two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

6. Select the appropriate items, then select **Move Items**. When you have added all the devices, click **Finish**. The selections appear in the list. To view programming options, double-click the extension number.

To delete a device from an attendant list, select the item, right-click, and select **Move to NONE List**.

You can also assign attendants in phone programming by selecting the phone Associated Extensions and Attendant options, and then selecting the phone that will serve as the attendant (see **page 392**).

PRIMARY ATTENDANTS

When you select Primary Attendants from the Phone-Related Programming list, you see fields for programming the Primary Attendant phone and the Local Attendant phone. Those fields are programmed as described below.

- **Primary Attendant**: This is the network primary attendant.

- **Local Attendant**: This is the attendant for the node being programmed.

If an attendant is set to None, calls that would normally go to the attendant are handled as follows:

- If the system has seized the call, but it has not been sent to a phone, the call is disconnected.

- If the call has been sent to a phone, it remains at the phone and rings until answered.

- If the call is not seized and not sent to a phone, the caller will hear ringing until he or she hangs up. The call will not ring at any phone.

To program an attendant:

1. Select System – Phone-Related Information – **Primary Attendant**.

2. Select the extension number.

3. Select one of the following methods:

   Method A

   a. In the **Value** column, select the current value, and then enter the new value in the box.

   b. Press **ENTER**. A screen appears displaying what is associated with the number entered.

   c. Click **OK**. The new number appears in the field.

   Method B

   a. Right-click the existing Primary Attendant Phone or Local Attendant Phone, and then click **Change Attendant**. A dialog box appears prompting for the device type to include.

   b. Select the desired device or **None**, and then click **Next**. The list of devices with details appears. To view items in a list only, click **List**.

   c. Select the desired device, and then click **Finish**. The selection appears in the applicable Attendant field.

SINGLE LINE PHONE CLID TIMERS

Single line phones in European and Australian systems only. The system supports the transmission of CLID to single line sets in Europe and Australia. This feature uses the calling party information that the system receives from the local network provider. Once programmed, on-hook single line phones display the calling party information when receiving an incoming outside call. The CLID information is also displayed if the single line set receives a transferred call from another phone that has calling party information. CLID is not transmitted to Single-Line Adapters (SLAs).

To use CLID for single line phones:

Enable the Caller ID flag as described on **page 419**.

You must also program the following timers:

- *Caller ID Alerting Tone:* To set this timer:

    a. Click the number in the **Value** column.

    b. Enter the number of milliseconds that the system will send Caller ID [CLID] alert tones to the single line phone. These tones are not audible to the end user and are used strictly to notify the phone that information will be transmitted. This value may differ for various phones. The valid range is 88–110 ms, and the default is 100 ms.

- *Caller ID Line Reversal*: To set this timer:

    a. Click the number in the **Value** column.

    b. Enter the number of milliseconds that the system reverses the line polarity so that the single line phone can receive Caller ID [CLID] information. This value may differ for various phones. The valid range is 0–200 ms, and the default is 30 ms.

NOTE CLIP is not transmitted to Single-Line Adapters (SLAs).

TROUBLESHOOTING PHONES

This section contains the troubleshooting tips for the following issues:

- **Phones** below

- PKM 16 on **page 490**

- Single Line Phones on **page 491**

- Multi-Protocol Phones on **page 494**

PHONES

Table 76 summarizes troubleshooting strategies recommended for resolving phone discrepancies.

Table 76:   Phone Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Phone inoperative; LED indication present while any button with an LED is held down; reorder tone is heard when button is pressed.	System lockout caused by excessive data errors (displays SYSTEM LOCKOUT).	Remove and replace the line cord to reset the phone.
	Programming error (circuit identified as dual single line sets—SLA; no reorder tone is heard)	Identify the circuit for phone use, not dual single line sets (SLA).
	Defective cabling or connections	Ensure that all cables are correctly connected to the modular jack as shown in the Installation chapter in the *MiVoice Office 250 Installation Manual* . Check for loose or open connections in the phone cabling and the line cord.
	Defective phone	Replace the phone if faulty.
	Defective Digital Endpoint Module	Replace the associated module.
	Defective processor module	Replace the module if faulty.
	Programming error or Improper configuration of IP connections, firewall and/or network address translation	Make sure the IP phone is programmed on the same subnet. See Appendices A and B in the *MiVoice Office 250 Installation Manual* for configuration guidelines.
	Older version or incompatible firmware on the phone	Verify the correct firmware and upgrade to the latest version through the phone Web interface. Reset the Web page to check for the latest version of firmware installed on the phone.
	IP resources are not allocated correctly; Display shows "No IP Resources Allocated"	Make sure you have enough IP resources and they are allocated correctly.

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Table 76:   Phone Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
	No IP connection and phone will continually reboot and recycle	Verify the LAN link light is active on the IP phone.
	Defective switch port, cabling, or improper power supply for the phone and phone will continually reboot and recycle	See if you can ping the system and check for network connectivity. Replace the faulty switch port, cabling, or install the correct power supply.
Phone inoperative; no LED indication when any button is pressed; no audio is present (continued)	Defective phone	Perform the phone self-test as described in the Installation chapter in the *MiVoice Office 250 Installation Manual* , and replace the phone if faulty.
	Defective cabling or connections	Ensure that 24/48 VDC is present at the power supply or the Power Over Ethernet (POE) switch and is correct. Check for loose or open connections in the phone cabling. For more information about hardware connections, refer to the Installation chapter in the *MiVoice Office 250 Installation Manual* .
	Defective power supply	Verify the phone has the correct power supply voltage.
	Defective Digital Endpoint Module	Replace the associated module.

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Table 76:   Phone Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Erratic phone operation (lamp status incorrect)	Phone cable exposed to interference	Ensure proper phone cable runs. For more information about cable runs, refer to the Installation chapter in the *MiVoice Office 250 Installation Manual* .
	Loop limits exceeded	Perform the phone loop resistance test as out lined in the Installation chapter in the *MiVoice Office 250 Installation Manual* .
	Defective cabling or connections	Ensure that 24/48VDC is present at the power supply or the Power Over Ethernet (POE) switch and is correct. Check for loose or open connections in the phone cabling. For more information, refer to the Installation chapter in the *MiVoice Office 250 Installation Manual* .
	Defective phone	Perform the phone self-test as described in the Installation chapter in the *MiVoice Office 250 Installation Manual* . Replace the phone if faulty.
	Programming error	Ensure that the proper keymap has been assigned to the phone. See "Keymaps" on **page 425**.
	Defective Digital Endpoint Module	Replace the associated module.
	Defective processor module	Replace the module if faulty.
	Database corruption	Run the database through the Test Utility.
	System memory is low	Run the diagnostics heap statistics to determine if the system is losing heap memory and how quickly the system may be losing it.
Phone squeals on outside calls or when receiving a handsfree intercom call from a single line phone (feedback)	Speaker volume is too loud	Reduce feedback by lowering speaker volume using phone volume controls.
	Poor acoustics	Poor acoustics can cause poor quality on handsfree calls. Try placing a private call through the handset.

Note: A two- to four-wire converter is used during communication between phones and single line stations. Reflection is a normal characteristic of these converters. Feedback on intercom calls is eliminated when the single line station user places a private intercom call by pressing the pound/hash (#) button before dialing the intercom number or by entering the ring intercom always feature code – 377 (provided the phone user does not press the SPKR button to respond).

Cannot break CO dial tone **Note:** See also CO trunk problems.	Defective phone	Perform the phone self-test as described in the Installation chapter in the *MiVoice Office 250 Installation Manual* .
	Defective trunk module	Replace defective trunk module.
	Defective Digital Endpoint Module	Replace the associated module.
	Defective processor module	Replace the module if faulty.

Table 76:   Phone Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Cannot obtain intercom dial tone; no tone heard; CO trunk works	Defective phone	Perform the phone self-test as described in the Installation chapter in the *MiVoice Office 250 Installation Manual* . Replace the phone if faulty.
	Defective Digital Endpoint Module	Replace the associated module.
	Defective processor module	Replace the module if faulty.
Cannot break intercom dial tone	Defective phone	Perform the phone self-test as described in the Installation chapter in the *MiVoice Office 250 Installation Manual* . Replace the phone if faulty.
	Defective Digital Endpoint Module	Replace the associated module.
	Defective processor module	Replace the module if faulty.
Cannot place intercom call; IC dial tone is present, but reorder tone is heard when call is attempted	User error	Invalid intercom number or improper dialing procedure.
	Defective phone	Perform the phone self-test as described in the Installation chapter in the *MiVoice Office 250 Installation Manual* . Replace the phone if faulty.
	Defective Digital Endpoint Module	Replace the associated module.
	Defective processor module	Replace the module if faulty.

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Table 76:   Phone Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Cannot receive Off-Hook Voice Announce (OHVA) calls	Programming error	Ensure that the system option for OHVA is enabled—see "System Flags" on **page 751**—that the called phone is programmed to receive OHVA calls, and that the calling phone is programmed to transmit OHVA calls—see the "Phone/Phantom/Hot Desk Profile Flags" on **page 417**.
	User error	The called phone is a single-line set or a phone that is programmed not to receive OHVA calls. Or, the called phone user may have blocked the OHVA call. For more information about OHVA, see "Off-Hook Voice Announce (OHVA)" on **page 323**.
	Secondary voice path busy	The phone does not receive off-hook voice announce calls when the secondary voice path or speakerphone are in use. For more information about OHVA, see "Off-Hook Voice Announce (OHVA)" on **page 323**.
	Defective PCDPM Module	Check installation and connections, and replace PCDPM if defective. For more information about PCDPM installation requirements, refer to the Installation chapter in the *MiVoice Office 250 Installation Manual* .
	Defective phone	Perform the phone self-test as described in the Installation chapter in the *MiVoice Office 250 Installation Manual* . Replace the phone if faulty.
	Defective Digital Endpoint Module	Replace the associated module.
	Defective processor module	Replace the module if faulty.

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Table 76:   Phone Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Cannot place off-hook voice announce calls	Programming error	Ensure that the system option for OHVA is enabled—see "System Flags" on **page 751**—that the called phone is programmed to receive OHVA calls, and that the calling phone is programmed to transmit OHVA calls—see the "Phone/Phantom/Hot Desk Profile Flags" on **page 417**.
	User error	The called phone is a single line set or a phone that is programmed not to receive OHVA calls. Or, the called phone user may have blocked the OHVA call.
	Secondary voice path busy	The called phone cannot receive off-hook voice announce calls when its secondary voice path or speakerphone are in use.
	Defective phone	Perform the phone self-test as described in the Installation chapter in the *MiVoice Office 250 Installation Manual* . Replace the phone if faulty.
	Defective Digital Endpoint Module	Replace the associated module.
	Defective processor module	Replace the module if faulty.
Data device connected to phone not operating properly	User error	Refer to manufacturer's operating instructions.
	Secondary voice path busy	The phone will not transmit data calls when the secondary voice path or speakerphone are in use.
	Problem with data device	Disconnect data device and check operation according to the manufacturer's instructions.
	PCDPM, MDPM, or AC transformer not installed properly or defective	Check installation and connections. For more information about hardware connections, refer to the Installation chapter in the *MiVoice Office 250 Installation Manual* .
	Data Port Module not installed properly or defective	Check Data Port Module installation and jumper strap settings. For more information about the Data Port Module, refer to the Installation chapter in the *MiVoice Office 250 Installation Manual* . Replace if defective.
	Defective phone	Perform the phone self-test as described in the Installation chapter in the *MiVoice Office 250 Installation Manual* . Replace the phone if faulty.
	Defective Digital Endpoint Module	Replace the associated module.
	Defective processor module	Replace the module if faulty.

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Table 76:   Phone Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Data noise in any off-hook condition	Defective cabling or connections (for example, line cord, amphenol connector, bridging clip, and so forth)	Check for loose or open connections, or crossed wires.
	Defective phone	Try another phone.
	Defective digital module	Replace the associated module.
Optional headset inoperative	User error	Ensure the enable headset feature code (315) was entered. Check feature code programming to see if code was changed. See "Feature Codes" on **page 766**.
	Incorrect or defective head set	Ensure the headset contains the appropriate microphone (dynamic or electret). Ensure the headset's configuration dial is set to where you can hear the tone (refer to the manufacturer's installation sheet). Replace headset if necessary.
	Defective phone	Try another phone.
	Defective processor module	Replace the module if faulty.
	Defective power in headset.	Replace the power or batteries. Also verify the manufacturer's settings for correct operation of the headset.
Delayed audio connection on incoming calls when using a headset with an Attendant Console computer	Headset in battery saving mode	Some headsets have a battery saving feature that turns the headset off after a certain period of inactivity. To prevent these headsets from missing portions of incoming calls, the phone flag can be programmed to generate a "Send Alert Burst To Headset" that activates the headset before connecting to an incoming call. See the "Phone/ Phantom/Hot Desk Profile Flags" on **page 417**. Or, if necessary, disable the feature and/or replace the batteries with a transformer.
	User error	Ensure the enable headset feature code (315) was entered. Check feature code programming to see if code was changed.
	Incorrect or defective head set	Ensure the headset contains the appropriate microphone (dynamic or electret). Ensure the headset's configuration dial is set to where you can hear the tone (refer to the manufacturer's installation sheet). Replace headset if necessary.
	Defective phone	Try another phone.
	Defective processor module	Replace the module if faulty.

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Table 76:   Phone Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
AgentSet inoperative	Not in headset mode	Ensure the enable headset feature code (319) was entered. Check feature code programming to see if code was changed. At the phone, unplug the headset or handset (to take the AgentSet off hook) and enter the Headset On/Off feature code (319) to enable headset mode. Then plug in the headset or handset.
	Incompatible headset	Use only a Mitel headset.
LRA device connected to analog phone not operating properly	Problem with LRA device	Disconnect LRA device and check operation according to the manufacturer's instructions.
	Data Port Module not installed properly or defective	Check Data Port Module installation and jumper strap settings. For more information about installing the Data Port Module, refer to the Installation chapter in the *MiVoice Office 250 Installation Manual* . Replace if defective.
	Defective analog phone	Try another phone.
	Programming error.	Make sure the Extended Ring Cadence flag is enabled.

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PKM 16

Table 77 summarizes the troubleshooting strategies recommended for resolving operational discrepancies with PKM 16 (Mini-DSS).

Table 77:   PKM 16 Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
PKM 16 unit inoperative; no LED indication present while button is pressed	Improper installation	Check for loose connections. For more information about hardware connections, refer to the Installation chapter in the *MiVoice Office 250 Installation Manual* .
	Defective PKM 16 unit	Replace the PKM 16 unit.
		Phone is identified for phone use, not as a PKM 16 unit. See "Programming DSS/PKM Keymaps" on **page 440**.
	The phone is not an 8660	Make sure you are using an 8660 phone.
	The phone is not an 8520 or 8560	Make sure you are using an 8520 or 8560 digital telephone.
	Programming error	Phone is identified for phone use, not as a PKM 16 unit. See "Programming DSS/PKM Keymaps" on **page 440**.

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Table 77: PKM 16 Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
PKM 16 unit inoperative; LED indication present while button is pressed	Programming error	Make sure the PKM 16 unit is enabled. See "Programming DSS/PKM Keymaps" on **page 440**.
	Defective PKM 16 unit	Replace the PKM 16 unit.
PKM 16 LED indications incorrect	Programming error	Check PKM 16 button assignments and make sure the phone is enabled as a PKM 16 unit. See "Programming DSS/PKM Keymaps" on **page 440**.
	Defective PKM 16 unit	Replace the PKM 16 unit.
	Defective processor module	Replace the module if faulty.
Calls are transferred to the wrong phone	User error	Refer to the applicable phone user guide for feature instructions.
	Programming error	Check PKM 16 button assignments. See "Programming DSS/PKM Keymaps" on **page 440**.

SINGLE LINE PHONES

Table 78 summarizes the troubleshooting strategies recommended for resolving discrepancies occurring with single line phones.

Table 78: Single Line Phone Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Single line phone does not work	Defective phone	Replace the single line phone.
	Defective cabling	Check the phone and system cabling. For more information about hardware connections, refer to the Installation chapter in the *MiVoice Office 250 Installation Manual* .
	Defective SLA	Replace the associated SLA.
	Defective Digital Endpoint Module	Replace the associated module.
	Defective processor module	Replace the module if faulty.
	Defective onboard single line ports	Replace the chassis.
Single line phone does not work; talk battery present	Defective set	Replace the single line phone.
	Defective DEM-16 or Single Line Module	Replace the associated defective module.
	Defective processor module	Replace the module if faulty.
	Defective onboard single line ports	Replace the chassis.

Table 78:   Single Line Phone Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Single line phone does not work; calls ring in; talk battery is present	Defective phone	Replace the single line phone.
	Defective SLA	Replace the associated SLA.
	Defective Digital Endpoint Module	Replace the associated module
	Defective processor module	Replace the module if faulty.
	Defective onboard single line ports	Replace the chassis.
A group of single line phones is inoperative; all are on the same SLA or the T1M or T1M-2 expansion module	Defective cabling	Ensure connector or cable is securely attached to the SLA or the T1M or T1M-2 expansion module.
	Defective SLA, T1M, or T1M-2 expansion module	Replace the module.
Single line phone will not ring for CO or intercom calls; talk battery is present; calls can be placed	Single line phone ring jumper strap not set for correct ringing	Set the ring jumper strap of the single line phone for AC ringing. For more information, refer to the Installation chapter in the *MiVoice Office 250 Installation Manual* .
	Defective phone	Replace the phone.
	Defective SLA	Replace the associated SLA.
	Defective Digital Endpoint Module	Replace the associated module.
	Defective processor module	Replace the module if faulty.
	Defective onboard single line ports	Replace the chassis.
Ring trip is not provided to a group of single line phones; outgoing calls are not affected	Defective cabling of the onboard single line ports.	Using a voltmeter, measure the input at the block. For more information, refer to the Installation chapter in the *MiVoice Office 250 Installation Manual* .
	Proper voltage absent from SLA.	Defective connection or incorrect installation of SLA. Using a voltmeter, measure the voltage at the digital expansion block. For more information, refer to the Installation chapter in the *MiVoice Office 250 Installation Manual* .
	Proper voltage absent from Digital Endpoint Module	Defective connection or incorrect installation of the Digital Endpoint Module. Using a voltmeter, measure the voltage at the digital block.
	Defective digital module	Replace the associated module.
A group of AC ringer-equipped single-line sets will not ring for CO or intercom calls; talk battery is present; calls can be placed; all phones are on the same SLA or Digital Endpoint Module	Defective SLA	Replace the associated SLA.
	Defective Digital Endpoint Module	Replace the associated module.
	Defective processor module	Replace the module if faulty.
	Programming error	Make sure the Extended Ring Cadence flag is enabled.

Table 78:   Single Line Phone Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
No AC ringer-equipped single line phone rings for CO or intercom calls; talk battery present; calls can be placed	Defective processor module	Replace the module if faulty.
	Defective Digital Endpoint Module	Replace the associated module.
	Defective SLA	Replace the associated SLA.
	Programming error	Make sure the Extended Ring Cadence flag is enabled.
	Either a DTMF decoder, voice channel, or tone generator is not available	Single line phone user will hear busy tones when any of the necessary resources are not available. User may camp on.
Cannot obtain intercom dial tone; no sound is heard	Defective set	Replace the single line phone.
	Defective SLA	Replace the associated SLA
	Defective Digital Endpoint Module	Replace the associated Digital Endpoint Module.
	Defective processor module	Replace the module if faulty.
Cannot place intercom call; dial tone present, but reorder tone heard	User error	Try the call again. User may have dialed an invalid number.
	Defective set	Replace the single-line set.
	Defective processor module	Replace the module if faulty.
Cannot break CO dial tone **Note:** See also CO trunk problems.	Defective phone	Replace the single line phone.
	Defective SLA	Replace the associated SLA.
	Defective Digital Endpoint Module	Replace the associated Digital Endpoint Module.
	Defective processor module	Replace the module if faulty.
	Defective onboard single line ports	Replace the chassis.
Cannot place off-hook voice announce calls	Programming error	Ensure that the system option for off-hook voice announce is enabled that the phone is programmed to transmit OHVA calls. Also, Ensure the Ring Intercom Always feature is disabled.
	User error	Called phone is a single-line set or a phone that is not equipped with a secondary voice path.
	Defective processor module	Replace the module if faulty.
Single line phones not receiving audio message waiting indications	Programming error	Audible Message indication for SL Sets flag must be enabled in the database. See the "System Flags" on **page 751** chapter.
Audio distortion. Talker and/or listener hears brief, noticeable changes in volume that do not sound normal.	Some analog single line phones transmit at abnormally high levels. The talker may be speaking loudly.	Use an analog single line phone that meets industry-standard transmit levels. Reduce the volume of the incoming audio.

Table 78:   Single Line Phone Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Single-line set does not receive caller ID information.	Single line phone flags may be programmed incorrectly in DB programming.	Set the Caller ID-IC and Extended Ring Cadence flags to **Yes**.
	The single line device may not support the shortened number of an extension.	Replace the device with a single-line device that supports the shortened number of an extension number and the split ring of an IC call.
	The single line device may stop collecting data at the second IC ring burst.	

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MULTI-PROTOCOL PHONES

Table 79 summarizes the troubleshooting strategies recommended for resolving discrepancies that may occur with multi-protocol phones.

Table 79: Multi-Protocol Phone Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
The multi-protocol phone powers up and all the lamps remain lit permanently.	The phone software is corrupt	The multi-protocol phone must be recovered using the following steps: 1. Set up a computer on the same LAN as the phone. 2. Make sure the computer has a TFTP server and the correct software image file. 3. Rename the software image from 86xx_x_x_x.bin (where the _x_x_x represents the version number) to 86xx.bin. For example: — Rename 8600_1_1_5.bin to 8600.bin — Rename 8620_1_1_5.bin to 8620.bin (note that the 8620 and 8622 share the same binary image) — Rename 8662_1_1_5.bin to 8662.bin (note that the 8622 and 8622 share the same binary image) — Rename 8690_1_1_5.bin to 8690.bin 4. Point the TFTP server at the directory that contains this software image file. 5. Configure the IP address of the computer as 192.168.200.202. The phone starts pulling this file from the TFTP server. When finished, the phone resets on its own.
The 8622 or 8662 phones are not uploading new firmware images as specified in the TFTP configuration files. OR The TFTP configuration files have been modified with new image versions, but the 8622 or 8662 phones have not uploaded the new firmware.	The image strings have not been updated	Make sure that these strings have been updated. For example, From 1.01S, 1.0.31, 1.0.33, or 1.1.5 to 2.0.03: 86xx_image_ver: HWID 2.0.03 86xx_image_url: 192.168.200.202/86xx/86xx_2_0_03.bin **Note:** If upgrading an 8622 from 1.01S to 1.1.5, use the following image string: 8620_image_ver: 8620 HWID 1 1.1.5 For releases 2.0 and later: itphone_image_ver: 2.0.03 itphone_image_url: 192.168.200.202/itphone itphone_2_0_03.bin

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Table 79:   Multi-Protocol Phone Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
While installing the client application on the 8690, an error message indicating that the setup failed appears	The phone does not have sufficient memory resources or the phone has lost power before the upgrade is complete	The client application installation on the 8690 has failed. You must reformat the flash file system on the phone.
The Windows CE.Net stops during the firmware upgrade	This may be caused by using most of the system resources (processor and RAM) during an upgrade	Try doing one of the following: • Restart the Windows CE upgrade by going into Pending Upgrades. Windows CE will still show up as "Pending" because it has not completed copying, which results in a reset of the 8690. Press **OK** to restart the upgrade and eventually this causes the original upgrade time out and displays the dialog box to restart that upgrade. Press **No** to cancel the original upgrade and continue with the second download that is in progress (usually still erasing flash at this point). • Launch the 8690 manually by using a USB mouse. The USB mouse does not require a lot of memory in the CPU. With the USB mouse attached the user can perform some basic functions and still see the 8690 phone responding. The mouse should be attached before the upgrade is started. **Note:** When an upgrade is in progress, avoid using the 8690 phone.
The 8622 or 8662 phone flashes the error message ERROR WRONG HWID when attempting to download a new image	The image that the 8622 or 8662 phone is attempting to download does not support the phone's hardware.	Use a more recent image, preferably at or above the version shipped with the phone.
A multi-protocol phone flashes the error message ERROR WRONG VERSION when attempting to download a new image	The version number in the xxxx_image_ver string may not match up with the image's version.	Correct the xxxx_image_ver image version number or the image name (if pointing at the wrong image file).

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Table 79:   Multi-Protocol Phone Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION

The 8622 phone flashes the error message ERROR WRONG VERSION when attempting to download a new image.	The phone does not have the latest firmware	The 8622 phones running firmware version 1.01S displays the error ERROR WRONG VERSION when attempting to download new firmware. To upload to new firmware to the 8622, first use one of the following version strings to upgrade to either versions 1.0.33 or 1.1.5:
		
		   8620_image_ver: 8620 1.0.33
		
		or
		
		   8620_image_ver: 8620 HWID 1 1.1.5
		
		Once the newer firmware has been loaded on the 8622 phone, change the image version string back to one of the following:
		
		   8620_image_ver: 1.0.33
		
		or
		
		   8620_image_ver: HWID 1 1.1.5
		
		or
		
		   itphone_image_ver: 2.*v.w.* (the *v.w.* indicates the firmware version, such as 2.0.03)
		
		Note: After the firmware has been upgraded, the phone will again display the error ERROR WRONG VERSION every time it attempts to upgrade until the configuration version line is returned to the normal state noted above in either the TFTP file or the Web page (that is, "8620_image_ver: 1.0.33").

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Table 79:   Multi-Protocol Phone Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
When the multi-protocol phones reset, network connectively on the downlink (that is, the LAN port labeled "PC" on the back on the phone) is temporarily lost.	The phones generally reset for one of the following reasons: • Phone setting or firmware updates - the phones may require a reset for certain configuration changes to take place and will reset if the firmware is updated. • Telephony system updates - the phones will reset when the Call Processing system disconnects their connection for updates on the Call Processing side. The phone will continue to reset every several minutes until the telephony system recovers and the phone can re-connect. • Loss of network connectivity - the phones will reset when network congestion or loss of network connectivity prevents necessary keep-alive packets from being exchanged between Call Processing and the phone.	Workarounds for this issue are as follows: • Minimize phone resets due to system updates - Schedule telephony system and phone updates for non-peak hours when phone and network connectivity outages are less service affecting. • Isolate and eliminate any network issues - Isolate and eliminate any network congestion issues that may cause the phone to lose communication with Call Processing and thus reset. • For critical applications that cannot tolerate any interruptions in LAN service, use an external switch in place of the downlink port for network connectivity until resets due to external factors can be minimized. In the case of (3) loss of network connectivity, make sure that the network in question meets the minimum requirements specified in the *VoIP Data Network Requirements*, part no. 835.2885. You can find the manual on the Mitel eDocs Web site (http://edocs.mitel.com).

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Chapter 9

EXTENSION LISTS AND SYSTEM GROUPS

INTRODUCTION

This chapter includes programming instructions for MiVoice Office 250 groups and lists. Groups and lists can be composed of system users, trunks, or system features. This chapter covers the following system groups and lists:

- **Extension Lists** below

- CO Trunk Groups on **page 504**

- Node Trunk Groups on **page 548**

- Hunt Groups on **page 552**

- Network Groups on **page 588**

EXTENSION LISTS

This section contains the following information:

- **Feature Description** below

- Programming Extension Lists on **page 502**

FEATURE DESCRIPTION

An extension list is a group of intercom extensions or trunk group extensions. You can use extension lists when you use program features that use common lists. For example, a group of phones could be assigned to the same paging zone and have ring-in for the same trunk groups. With an extension list, you would have to enter only one list number instead of repeatedly entering all of the phone numbers for each list.

DEFAULT EXTENSION NUMBERS

Extension numbers are recognized as feature codes by the system. Default extension numbers appear in **Table 80**.

Table 80: Default Extension Numbers

APPLICATION	EXTENSION NUMBER OR RANGE OF NUMBERS
Attendant	0
Phones	1000–1999
Modem	1500
Hunt Groups	2000–2074 or 2000–2299
Voice Processing Applications	2500–2999
Page Zones	9600–9609 or 9600–9649 Used only for programming a feature button to select a page zone.
External Page Ports	91000–91003

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Table 80:   Default Extension Numbers (continued)

APPLICATION	EXTENSION NUMBER OR RANGE OF NUMBERS
Individual Trunks	94000–95999 Determined by module bay and device number. Used for programming and troubleshooting.

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TYPES OF EXTENSION LISTS

The following are Extension List types:

- **ACD Agent ID**: Includes only ACD hunt group Agent IDs. If no Agent IDs have been created, as described on **page 569**, this option cannot be used.

- **IP/Digital Telephone**: Includes only phone extension numbers. The Auto: All Phones (PP052) extension list is provided by default and cannot be deleted. This list contains all of the phones programmed and is automatically updated when phones are added/deleted.

- **Single Line**: Includes only single line phone and MDPM extension numbers.

- **Phone**: Can include phones, MDPMs, and/or single line extension numbers. The Auto: All Phones/Voice Processor Applications (PP051) extension list is provided by default and cannot be deleted. This list contains all of the phones programmed and is automatically updated when phones are added or deleted.

NOTE	You cannot change or delete the Auto: All Phones/Voice Processor Apps or Auto: All IP/Digital Telephones extension lists.

- **CO Trunk Group**: Includes only trunk group access numbers.

- **Hunt Group Members**: Includes only phones that are members of hunt groups.

Phone and Agent ID extension lists can be included in hunt groups. For example, a hunt group can be programmed to send calls first to an individual phone, then to an extension list where all phones on the list will be rung, and then to another individual phone or an extension list.

The number of entries in individual lists limits the total number of extension lists allowed on the system. In all of the extension lists combined, a maximum of 2500 phone or trunk extensions entries is allowed.

AUTOMATIC EXTENSION LISTS

NOTE	When using an extension list for ring-in or hunt groups, do not exceed 30 phones per list. The system can send ring signal to up to 30 phones. When using a PS-1, up to 60 phones can be on the extension list.

In addition to the programmable extension lists, the system software supports two automatic extension lists:

- PP051: Auto: All Phones/Voice Processor Applications

- PP052: Auto: All IP/Digital Telephones

When a phone or voice mail application is equipped or unequipped, the system automatically adds or removes that extension in the appropriate list(s). You cannot delete or renumber the

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auto extension lists. You can only add or remove extensions from the lists in online monitor mode (not recommended).

The two auto extension lists default with all existing phones and voice mail applications. In a default database, PP051: Auto: All Phones/Voice Processor Applications is assigned to Day/Night Outgoing Access and Emergency Outgoing Access for the following groups:

- CO Trunk Groups

- Node Trunk Groups

- Node IP Connection Groups

PROGRAMMING EXTENSION LISTS

This section contains the following information:

- **Viewing Extension Lists** below

- **Creating Extension Lists** below

- Adding Devices to Extension Lists on **page 503**

- Deleting Extension Lists on **page 503**

VIEWING EXTENSION LISTS

To view the available lists:

1. Select System – Devices and Feature Codes – **Extension Lists** – <*extension list type*>.

2. Select the extension list.

CREATING EXTENSION LISTS

> **NOTICE**
>
> Adding a large extension list can result in slowing down system performance. Adding an Extension List that contains more than 60 members may cause a system slowdown because when the list is called, ALL members of the list are called at the same time.
>
> When using an extension list for ring-in or hunt groups, do not exceed 30 phones per list. The system can send ring signal to up to 30 phones (60 phones with a PS-1).
>
> If you exceed the number of phones and the system suffers a slowdown, Mitel Technical Support advises that you reduce the number of extension list members.

The number of entries in individual lists limits the total number of extension lists allowed on the system. In all of the extension lists combined, a maximum of 2500 phone or trunk extensions entries is allowed.

To create an extension list:

1. Select System – Devices and Feature Codes – **Extension Lists**.

2. Select the type of list you want to create.

3. Right click in the right pane, and then click **Create <*type*> Extension List**. The Create Extension List dialog box appears.

4. In the **Starting Extension** box, select or enter the starting extension for the new list.

5. In the **Number of Extensions** box, select or enter the number of extensions that you want to add.

6. Click **OK**.

7. The new list automatically appears in the list for that extension list type.

8. Select **Description**, and type the name, up to 20 characters, in the box.

9. When finished, press **ENTER** or click out of the field to save the change.

ADDING DEVICES TO EXTENSION LISTS

You can add devices to extension lists.

To program a list:

1. Select System – Devices and Feature Codes – Extension Lists – <***extension list type***>.

2. Select the extension list.

3. Right-click anywhere in the right pane, and then click **Add To List**. A window appears prompting for the device types to include.

4. Select the device types (you can use the SHIFT or CTRL key to select more than one item), and then click **Next**. The devices with details appear. To view devices in a list only, click **List**.

5. Select the devices you want to add to the list, and then click **Finish**. The selection appears in the extension list window.

DELETING EXTENSION LISTS

To delete an extension list:

1. Select System – Devices and Feature Codes – Extension Lists – <***extension list type***>.

2. Select the extension list.

3. Right-click and select Delete. The extension list is removed.

CO TRUNK GROUPS

Central Office (CO) trunk groups bundle system trunks into one group, or trunk group. Whenever a call is placed from a trunk in a CO trunk group, the system hunts for an available trunk to route the call out to the CO. Calls placed to CO trunk groups are routed according to the first available trunk.

Because MGCP and SIP trunks are CO trunks, they can also be in a CO Trunk group and need the applicable IP resources to be able to place calls. For more information, see "SIP Gateways" on **page 230** or "MGCP Gateways, Devices, and Trunks" on **page 234**. 92001 is the default baseline extension used for CO Trunk Groups; PP011 the default CO trunk group for unused trunk groups. You can view CO trunk groups programmed for the local node, as shown in the figure below.
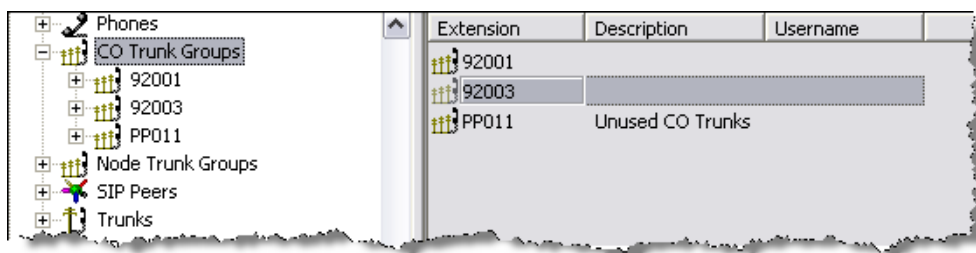
To view CO trunk groups:

Select System – Devices and Feature Codes – **CO Trunk Groups**.

This section contains the following information:

- **Viewing Trunks in a Trunk Group** below
- Adding CO Trunk Groups on **page 505**
- Changing CO Trunk Group Extension Numbers on **page 505**
- Moving Trunks Between CO Trunk Groups on **page 506**
- Programming Trunk Group Options on **page 506**
- Trunk Camp On and Busy Trunk Queue Callback on **page 526**
- Toll Restriction on **page 527**
- Caller ID Forwarding on **page 534**

VIEWING TRUNKS IN A TRUNK GROUP

Individual trunks are assigned to trunk groups. You can view and edit the trunks in each trunk group.

To view the trunks assigned to a trunk group:

Select System – Devices and Feature Codes – CO Trunk Groups – *<trunk group number>* – **Trunks**.

ADDING CO TRUNK GROUPS

You can add CO trunk groups.

To add a CO trunk group:

1. Select System – Devices and Feature Codes – **CO Trunk Groups**.

2. Right-click in the right pane, and then select **Create CO Trunk Group**. The Create CO Trunk Group Extension dialog box appears.

3. Enter the starting extension and number of extensions for the new trunk group, and then click **OK**.

4. Program the **Description** and **Username** fields. All trunk groups should have a description and a username. The description appears in all trunk group lists in the database and can be up to 20 characters long. The username appears on phone displays and can have up to 10 characters.

   To program the Description and Username fields:

   a. Select the box that you want to program.

   b. Type the entry.

   c. Press **ENTER** or click out of the field to save your changes.

CHANGING CO TRUNK GROUP EXTENSION NUMBERS

You can change either individual or multiple trunk group extension numbers.

To change an individual trunk group extension number:

1. Select System – Devices and Feature Codes – **CO Trunk Groups**.

2. Select the current extension number.

3. Select the new number from the list or type the number in the list box.

4. Press **ENTER** or click out of the box to save your changes.

To change multiple trunk group extension numbers at one time:

1. Select System – Devices and Feature Codes – **CO Trunk Groups**.

2. Select the trunk group extensions that you want to change. You can use the SHIFT or CTRL key to select more than one extension.

3. Right-click, and then select **Batch Extension Change**. The Create CO Trunk Group dialog box appears.

4. Select the number you want to assign to the first selected trunk group (the other trunk groups will be numbered consecutively after this number).

5. Click **OK**. The trunk groups are automatically renumbered and resorted in the phone list.

MOVING TRUNKS BETWEEN CO TRUNK GROUPS

You can move trunks between trunk groups.

To move trunks to from another trunk group into the selected trunk group:

1. Select System – Devices and Feature Codes – **CO Trunk Groups**.

2. Select the trunk group number.

3. Double-click **Trunks**, and then right-click in a blank area in the right pane.

4. Click **Move To Trunks List**. A window appears prompting for the device type to include.

5. Select the trunk types, and then click **Next**. A list of available trunks with details appears. To view trunks in a list only, click **List**.

6. Select the appropriate trunks, and then select **Move Items**.

7. When you have added all the desired trunks, click **Finish**. The selections appear in the list.

To move trunks out of the selected trunk group to another trunk group:

1. Select System – Devices and Feature Codes – CO Trunk Groups – <**trunk group**>.

2. Double-click **Trunks**.

3. Drag and drop the trunks into the trunk list of the new trunk group. Use the **CTRL** or **SHIFT** keys to select several trunks at a time.

PROGRAMMING TRUNK GROUP OPTIONS

You can program the following CO or SIP peer trunk group configuration options. To assign trunks to CO trunk groups, see "Assigning Trunks to CO Trunk Groups" on **page 220**. For details about SIP trunk groups, see "Service Provider SIP Trunks and SIP Trunk Groups" on **page 691**.

Table 81:   CO and SIP Peer Trunk Group Programming Options

OPTION \ TRUNK GOUP	REF. PAGE	CO TRUNK GROUPS	SIP TRUNK GROUPS
Day or Night Multiple Ring-In Types	**508**	✓	✓
Day or Night Answer Access	**508**	✓	
Day or Night Emergency Outgoing Access	**509**	✓	✓

Table 81:   CO and SIP Peer Trunk Group Programming Options (continued)

OPTION \ TRUNK GOUP	REF. PAGE	CO TRUNK GROUPS	SIP TRUNK GROUPS
Day or Night Outgoing Access	**510**	✓	✓
Toll Restrictions	**511**	✓	✓
Search Algorithm	**513**	✓	
Audio for Calls Camped onto this Device	**514**	✓	✓
Music-On-Hold	**514**	✓	✓
Audio on Transfer To Ring	**515**	✓	✓
Audio On Transfer To Hold	**515**	✓	✓
Audio On Hold For Transfer Announcement	**516**	✓	✓
Audio for Calls Holding for this Device	**517**	✓	✓
PRI Call By Call Service	**517**	✓	
One-Way Incoming Only	**518**	✓	
Echo Trunk Number	**518**	✓	✓
Enable Hookflash	**519**	✓	
Camp-Ons Allowed	**519**	✓	
ISDN Data Calls Allowed	**520**	✓	
Day and Night Ring-In Types	**520**	✓	✓
Send Station Extension/Username to Attached PBX	**522**	✓	✓
Propagate Original Caller ID	**523**	✓	✓
Do Not Propagate Original Caller ID to P-Asserted Identity	**524**	✓	✓
Calling Party Name	**525**	✓	✓
Calling Party Number	**525**	✓	✓
Force Trunk Group Calling Party Name and Number	**525**	✓	✓
Wait for ISDN Caller ID Information	**525**	✓	

DAY OR NIGHT MULTIPLE RING-IN TYPES

If the Ring-In Type (see **page 520**) is set to "Multiple," you can determine the ring-in destinations for day and night modes. The trunk group can ring in to a list of phones, extension lists, or applications, (but not hunt groups). The list can include local or off-node device extension numbers. When using an extension list for ring-in, do not exceed 30 phones for each list.

To set the ring-in destination:

1. Do one of the following:

   • Select System – Devices and Feature Codes – **CO Trunk Groups**, and then select the trunk group number.

   • System – Devices and Feature Codes – SIP Peers – SIP Trunk Groups – *<SIP trunk group>* – **Trunk Group Configuration**.

2. Double-click **Multiple Ring-In**, and then double-click **Day** or **Night**.

3. Add destinations to the list:

   a. Right-click anywhere in the right pane, and then click **Add to Day** (or **Night**) **List**. The Add to Day (or Night) dialog box appears.

   b. Select the device types (you can use the SHIFT or CTRL key to select more than one item), and then click **Next**. The items with details appear. To view items in a list only, click **List**.

   c. Select the appropriate items, and then click **Add Items**.

   d. Click **Finish**. The selections appear in the list. To view programming options, double-click the extension number.

To delete destinations from the list:

1. Do one of the following:

   • Select System – Devices and Feature Codes – **CO Trunk Groups**, and then select the trunk group number.

   • System – Devices and Feature Codes – SIP Peers – SIP Trunk Groups – *<SIP trunk group>* – **Trunk Group Configuration**.

2. Select **Multiple Ring-In**, and then select **Day** or **Night**.

3. Select the item, right-click, and then select **Remove Selected Items**.

DAY OR NIGHT ANSWER ACCESS

This feature is not available for DISA trunks. Day or Night Answer Access allows phone users to answer incoming calls on the trunks in that trunk group (even if the phone does not have ring-in assignment for that trunk group). Phones cannot have allowed-answer assignment for trunk groups on other nodes.

To program trunk-group extensions for allowed-answer permission in day or night mode:

1. Select System – Devices and Feature Codes – **CO Trunk Groups**.

2. Select the trunk group number.

3. Select **Answer Access**, and then select **Day Mode** or **Night Mode**.

4. To add extensions to the list:

   a. Right-click anywhere in the right side of the window. An option box appears.

   b. Select **Add To List**. A window appears prompting for the device type to include.

   c. Select the device types (you can use the SHIFT or CTRL key to select more than one item), and then click **Next**. The items with details appear. To view items in a list only, click **List**.

   d. Select the appropriate items, then select **Add Items**. When you have added all the desired devices, click **Finish**. The selections appear in the list. To view programming options, double-click the extension number.

To delete destinations from the list:

1. Select System – Devices and Feature Codes – **CO Trunk Groups**.

2. Select the trunk group number.

3. Select **Answer Access**, and then select **Day Mode** or **Night Mode.**

4. Select the extension number.

5. Right-click, and then select **Remove Selected Items**.

DAY OR NIGHT EMERGENCY OUTGOING ACCESS

You can program emergency outgoing access for day or night mode. For loop start trunks that are connected to paging equipment, the emergency outgoing access designation does *not* keep the phone from seizing the trunk for paging. Trunks used for paging should not allow any phone to have emergency outgoing access. By default, the automatic phone list (Auto: All Phones) is assigned to Day/Night Emergency Outgoing Access.

To add phones with emergency outgoing access for the trunk group:

1. Do one of the following:

   • Select System – Devices and Feature Codes – **CO Trunk Groups**, and then select the trunk group number.

   • System – Devices and Feature Codes – SIP Peers – SIP Trunk Groups – *<SIP trunk group>* – **Trunk Group Configuration**.

2. Select **Emergency Outgoing Access**, and then select either **Day Mode** or **Night Mode**.

3. Right-click anywhere in the right side of the window. An option box appears.

4. Select **Add To List**. A window appears prompting for the device type to include.

5. Select the device types (you can use the SHIFT or CTRL key to select more than one item), and then click **Next**. The items with details appear. To view items in a list only, click **List**.

6. Select the appropriate items, then select **Add Items**. When you have added all the desired devices, click **Finish**. The selections appear in the list. To view programming options, double-click the extension number.

To delete destinations from the list:

1. Select **Emergency Outgoing Access**, then select either **Day Mode** or **Night Mode**.

2. Select the item, right-click, and then select **Remove Selected Items**.

> ## NOTICE
>
> Responsibility for Regulatory Compliance.
>
> It is the responsibility of the organization and person(s) performing the installation and maintenance of Mitel Advanced Communications Platforms to know and comply with all regulations required for ensuring Emergency Outgoing Access at the location of both the main system and any remote communication phones. Remote IP and SIP phones may require gateway access to nearby emergency responders.
>
> Emergency Call phone numbers include:
>
> - 911, the default for Mitel systems located in the U.S.
> - 999, the default for Mitel systems located in the European market and used primarily in the U.K.
> - If applicable, 112, an emergency number used widely in Europe outside of the U.K.
> - 112, the default for Mitel systems located in Australia.
> - Any emergency number, such as for a police or fire station, that is appropriate for the location of the main system and/or remote phones.

DAY OR NIGHT OUTGOING ACCESS

There are separate lists for phones with outgoing access in day and night modes. On loop start trunks that are connected to paging equipment, the outgoing access designation does not keep the phone from seizing the trunk for paging. Trunks used for paging should not allow any phone to have outgoing access for placing outside calls. By default, the automatic phone list (Auto: All Phones) is assigned to Day/Night Outgoing Access when a CO trunk group is created.

To add phones that will have outgoing access for the trunk group:

1. Do one of the following:
   - Select System – Devices and Feature Codes – **CO Trunk Groups**, and then select the trunk group number.
   - System – Devices and Feature Codes – SIP Peers – SIP Trunk Groups – *<SIP trunk group>* – **Trunk Group Configuration**.

2. Select **Outgoing Access**, and then select either **Day Mode** or **Night Mode**.

3. Right-click anywhere in the right side of the window. An option box appears.

4. Select **Add To List**. A window appears prompting for the device type to include.

5. Select the device types (you can use the SHIFT or CTRL key to select more than one item), and then click **Next**. The items with details appear. To view items in a list only, click **List**.

6. Select the appropriate items, then select **Add Items**. When you have added all the desired devices, click **Finish**. The selections appear in the list. To view programming options, double-click the extension number.

To delete destinations from the list:

1. Do one of the following:

- Select System – Devices and Feature Codes – **CO Trunk Groups**, and then select the trunk group number.

- System – Devices and Feature Codes – SIP Peers – SIP Trunk Groups – *<SIP trunk group>* – **Trunk Group Configuration**.

2. Select **Outgoing Access**, and then select either **Day Mode** or **Night Mode**.

3. Select the item, right-click, and then click **Remove Selected Items**.

TOLL RESTRICTIONS

You can program toll restrictions for each trunk group. For more information about toll restrictions and classes of service, see **page 527**.

To program toll restriction for the trunk group:

1. Do one of the following:

- Select System – Devices and Feature Codes – **CO Trunk Groups**, and then select the trunk group number.

- System – Devices and Feature Codes – SIP Peers – SIP Trunk Groups – *<SIP trunk group>* – **Trunk Group Configuration**.

2. Select **Toll Restriction**.

3. Program the following toll restriction parameters for each trunk group. For loop start trunks that are connected to paging equipment, toll restriction does not keep the phone from seizing the trunk for paging.

- **Class Of Service**: Each trunk group has Classes of Service (COS), which restrict or allow certain dialing patterns from being dialed on a call.

NOTE	When determining toll restriction for an ARS outgoing call, the network only checks the phone toll restriction for the node on which the calling phone resides. The system does not check the trunk class of service for ARS calls. To program the toll restrictions, double-click **Classes of Service**, you then have the option of choosing Day or Night. Double-click the desired time period to view a list of current classes of service.

To add a class of service:

a. Right-click in the window and select **Add To List**.

b. A window appears prompting for the toll restriction types you want to program. Select **ARS Only & Deny Area/Office** and/or **Classes of service**, and then click **Next**.

c. Another window appears displaying a list of available classes of service with details. To view items in a list only, click **List**.

d. Select the classes of service, and then select **Add Items**. The selected classes of service appear in the list. Click **Finish** to exit.

e. *(U.S. only) If you selected Deny Area/Office class of service,* the trunk group must also be assigned to a User Group. To change the user group, right-click **User Group** and select **Change User Group**. In the first window that appears, select **User Groups**, then click **Next**. In the next window, select the desired user group, and then click **Finish** to exit and save the change.

To delete one or more classes of service:

a. Select the items.

b. Right-click, and then select **Remove Selected Items**. To select a series of items, hold down SHIFT while selecting the first and last item in the range. To select two or more that are not consecutive, hold down CTRL while selecting the desired items.

- **Subject to Toll Restriction**: If the trunk group is not subject to toll restriction, the phone and trunk classes of service (except ARS-Only restriction) are not checked when a trunk in the group is used.

To enable this option:

a. Select the current value, and then select the check box. The field changes to **Yes**. To disable the flag, clear the check box.

b. Press **ENTER** or click out of the field to save your changes.

- **Exempt from ARS only**: The trunk group can be exempt from, or subject to, the ARS-only restriction. If exempt from ARS-only, phones with that restriction can directly select the trunks. If subject to ARS-only, ARS-restricted phones can use the trunk group only if it is part of an ARS route group.

To enable this option:

a. Select the current value, and then select the check box. The field changes to **Yes**. To disable the flag, clear the check box.

b. Press **ENTER** or click out of the field to save your changes.

- **Assigned Call Cost**: For trunk groups that are not subject to toll restriction, the call cost rate to be used for calls placed on the trunks can be free, local, toll local, toll long distance, operator-assisted, or international rate for the U.S. and free, local, toll (national), international, or operator rate for Europe.

To change the Call Cost setting:

a. Select the Assigned Call Cost value, then scroll to the desired setting.

b. Press **ENTER** or click outside the field to save the change.

- **Absorbed Digits**: *(For CO trunk groups only)* Trunk groups can be programmed to absorb digits in areas where the first digit(s) of the office code are absorbed. There can be up to 50 patterns with up to 48 digits each.

To enable the option:

a. Select the current value, and then select the check box. The field changes to **Yes**. To disable the flag, clear the check box.

    b. Press **ENTER** or click out of the field to save your changes.

- **Absorbed Patterns**: *(For CO trunk groups only)* If the Absorbed Digits and PBX Line flags are enabled, use this field to enter the absorbed digit patterns. There can be up to 50 patterns with up to 48 digits each. The only characters allowed in the pattern are digits (0-9), pound (or hash) sign (#), parentheses ( ), brackets ([ ]), greater than (>) and less than (<) symbols, and the wildcards. (See **page 368** for wildcard information.) Double-click **Absorbed Patterns** to view the current list of patterns, if any.

To add a pattern, do the following:

a. Right-click anywhere in the right side of the screen and select **Add To Absorbed Patterns List**. A new, blank pattern appears on the list.

b. Select the pattern to display the text box, then enter the desired pattern.

c. Press **ENTER** or click outside the field to save the change.

d. Repeat these steps if you need additional patterns.

- **PBX Line**: *(For CO trunk groups only)* Trunk groups can be programmed to absorb a digit string (defined above) for PBX installations.

To enable this option:

a. Select the current value, and then select the check box. The field changes to **Yes**. To disable the flag, clear the check box.

b. Press **ENTER** or click out of the field to save your changes.

- **Absorbed Digit String**: *(For CO trunk groups only)* If the PBX Line flag, described above, is enabled, use this field to enter the absorbed digit string.

To change the Absorbed Digit String:

a. Click the current value, then enter the new digit string, up to 32 digits.

b. Press **ENTER** or select another field to save the change. The only characters allowed in the string are digits (0–9), pound (or hash) sign (#), parentheses ( ), brackets ([ ]), greater than (>) and less than (<) symbols, and the wildcards. For more information about wildcards, see "Using the Wildcard Character in Off-Node Extensions" on **page 368**.

SEARCH ALGORITHM

You can program trunk groups to search trunk availability in either linear or distributed order:

- **Linear**: Requests for an outgoing trunk are always processed beginning with the highest numbered trunk on the list and moving down the list until an available trunk is found.

- **Distributed**: The first request will be processed beginning with the highest numbered trunk on the list. The next request will begin with the second trunk, and each subsequent request will begin one trunk lower on the list. When the end of the list is reached, requests begin again with the highest numbered trunk on the list.

To change the trunk group searching algorithm:

1. Select System – Devices and Feature Codes – **CO Trunk Groups**.

2. Select the trunk group number.

3. Select **Search Algorithm**.

4. In the **Value** column, select the setting.

5. Press **ENTER** or click outside the field to save the change.

AUDIO FOR CALLS CAMPED ONTO THIS DEVICE

You can select the audio that callers hear when camped-on to the trunk group.

To select the Audio for Calls Camped onto this Device:

1. Do one of the following:
   - Select System – Devices and Feature Codes – **CO Trunk Groups**, and then select the trunk group number.
   - System – Devices and Feature Codes – SIP Peers – SIP Trunk Groups – <*SIP trunk group*> – **Trunk Group Configuration**.

2. Select **Audio for Calls Camped onto this Device**.

3. In the **Value** column, select one of the following options from the list:
   - **Silence**: Callers hear no Music-On-Hold.
   - **Tick Tone**: Callers hear tick tone.
   - **Ringback**: Callers hear ringback.
   - **MiVoice Office 250**: Callers hear an external music source. This is the default value.
   - **File-Based MOH**: Callers hear the MOH file selected in DB Programming. For more information, see "File-Based Music-On-Hold (MOH)" on **page 741**.

4. Press **ENTER** or click outside the field to save the change.

MUSIC-ON-HOLD

You can select the audio that callers hear when placed on hold for the trunk group.

To select the Music-on-Hold audio:

1. Do one of the following:
   - Select System – Devices and Feature Codes – **CO Trunk Groups**, and then select the trunk group number.
   - System – Devices and Feature Codes – SIP Peers – SIP Trunk Groups – <*SIP trunk group*> – **Trunk Group Configuration**.

2. Select **Music-On-Hold**.

3. In the **Value** column, select one of the following options from the list:
   - **Silence**: Callers hear no Music-On-Hold.

- **Tick Tone**: Callers hear tick tone.
- **Ringback**: Callers hear ringback.
- **MOH Port**: Callers hear an external music source. This is the default value.
- **Use Next Device's Audio Source:** Callers hear the audio programmed at individual phones. See "Audio for Calls Holding for this Device" on **page 415**.
- **File-Based MOH**: Callers hear the MOH file selected in DB Programming. For more information, see "File-Based Music-On-Hold (MOH)" on **page 741**.

4. Press **ENTER** or click outside the field to save the change.

AUDIO ON TRANSFER TO RING

You can select the audio that outside callers (except those using DISA) hear while their call is being transferred. The audio that DISA callers hear is determined by how the DISA Transfer tone system flag is programmed (see **page 752**).

To select the Audio on Transfer to Ring:

1. Do one of the following:
   - Select System – Devices and Feature Codes – **CO Trunk Groups**, and then select the trunk group number.
   - System – Devices and Feature Codes – SIP Peers – SIP Trunk Groups – *<SIP trunk group>* – **Trunk Group Configuration**.

2. Select **Audio on Transfer to Ring**.

3. In the **Value** column, select one of the following options from the list:
   - **Silence**: Callers hear no Music-On-Hold.
   - **Tick Tone**: Callers hear tick tone.
   - **Ringback**: Callers hear ringback. This is the default value.
   - **MiVoice Office 250**: Callers hear an external music source.
   - **Music-On-Hold**: Callers hear the default system Music-On-Hold source.
   - **Use Next Device's Audio Source**: Callers hear the audio programmed at individual phones. See "Device Audio for Calls Settings" on **page 414**.
   - **File-Based MOH**: Callers hear the MOH file selected in DB Programming. For more information, see "File-Based Music-On-Hold (MOH)" on **page 741**.

4. Press **ENTER** or click outside the field to save the change.

AUDIO ON TRANSFER TO HOLD

You can select the audio that outside callers (except those using DISA) hear when they are transferred to and held at a different extension. The audio that DISA callers hear is determined by how the DISA Transfer tone system flag is programmed (see **page 752**).

To select the Audio on Transfer To Hold:

1. Do one of the following:

- Select System – Devices and Feature Codes – **CO Trunk Groups**, and then select the trunk group number.

- System – Devices and Feature Codes – SIP Peers – SIP Trunk Groups – <*SIP trunk group*> – **Trunk Group Configuration**.

2. Select **Audio on Transfer to Hold**.

3. In the **Value** column, select one of the following options from the list:

- **Silence**: Callers hear no Music-On-Hold.

- **Tick Tone**: Callers hear tick tone.

- **Ringback**: Callers hear ringback.

- **MiVoice Office 250**: Callers hear an external music source.

- **Music-On-Hold**: Callers hear the default Music-On-Hold source. This is the default value.

- **Use Next Device's Audio Source**: Callers hear the audio programmed at individual phones. See "Device Audio for Calls Settings" on **page 414**.

- **File-Based MOH**: Callers hear the MOH file selected in DB Programming. For more information, see "File-Based Music-On-Hold (MOH)" on **page 741**.

4. Press **ENTER** or click outside the field to save the change.

AUDIO ON HOLD FOR TRANSFER ANNOUNCEMENT

You can select the audio that outside callers (except those using DISA) hear while on transfer hold. After the transfer is completed, the caller hears the Audio On Hold For Transfer Announcement selection. The audio that DISA callers hear is determined by how the DISA Transfer tone system flag is programmed (see **"DISA Transfer Tone"** on **page 752**).

If the trunk group audio field, including Music-On-Hold, is set to Use Next Device's Audio Source, the system uses the programming for the next device as programmed for the Day/Night trunk group destination. If the field is set to any other option, the system uses the trunk group audio source, overriding phone programming.

To select the Audio On Hold For Transfer Announcement:

1. Do one of the following:

- Select System – Devices and Feature Codes – **CO Trunk Groups**, and then select the trunk group number.

- System – Devices and Feature Codes – SIP Peers – SIP Trunk Groups – <*SIP trunk group*> – **Trunk Group Configuration**.

2. Select **Audio On Hold for Transfer Announcement**.

3. In the Value column, select one of the following options from the list:

- **Silence**: Callers hear no Music-On-Hold.

- **Tick Tone**: Callers hear tick tone.

- **Ringback**: Callers hear ringback.

- **MiVoice Office 250**: Callers hear an external music source.

- **Music-On-Hold**: Callers hear the default Music-On-Hold source. This is the default value.

    - **Use Next Device's Audio Source**: Callers hear the audio programmed at individual phones. See "Device Audio for Calls Settings" on **page 414**.

        - **File-Based MOH**: Callers hear the MOH file selected in DB Programming. For more information, see "File-Based Music-On-Hold (MOH)" on **page 741**.

4. Press **ENTER** or click outside the field to save the change.

AUDIO FOR CALLS HOLDING FOR THIS DEVICE

This field is used whenever a trunk places a call on hold. The only time a trunk can place a call on hold is when using Mid Call Features (MCF).

> **NOTE** Mid-Call Features are only supported with other networked MiVoice Office 250s running Release v6.0 SP1 and higher software.

The device talking to the trunk will use this setting to configure the appropriate audio when on hold. The following options are available for use:

- Silence

- Tick Tone

- Ringback

- MiVoice Office 250

- Music on Hold

- Use Next Device's Audio Source

- File Based Music on Hold

> **NOTE** If the trunk group audio field, including Music-On-Hold, is set to Use Next Device's Audio Source, the system uses the programming for the next device as programmed for the Day/Night trunk group destination. If the field is set to any other option, the system uses the trunk group audio source, overriding phone programming.

> **NOTE** This field can also be used specify hold audio in those cases where an outgoing CO trunk is put on hold by the far end.

PRI CALL BY CALL SERVICE

This feature applies to U.S. installations only. If your trunk group uses PRI B-channels, you can select the PRI Call By Call service for outgoing calls. For a list of supported services, see "Trunk Group PRI Call By Call Feature" on **page 166**.

To select the PRI Call By Call Service:

1. Select System – Devices and Feature Codes – **CO Trunk Groups**.

2. Select the trunk group number.

3. Select **PRI Call By Call**, and then select the service from the list.

4. Press **ENTER** or click outside the field to save the change.

ONE-WAY INCOMING ONLY

You must designate certain types of trunks, such as incoming Wide Area Telecommunications Service (WATS), as "incoming-only," so the system recognizes the lines during power-up or testing. On these types of trunks, battery is not returned when the line is seized and the system cannot power up the trunk *unless* this option is enabled.

To program a trunk as One-Way Incoming Only:

1. Select System – Devices and Feature Codes – **CO Trunk Groups**.

2. Select the trunk group number.

3. Select **One-Way Incoming Only**.

4. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

> **NOTE** | When the check box is selected, the system checks to determine whether any phones have outgoing access for that trunk group. *If there are phones with outgoing access,* a message prompts you to delete the outgoing-access assignments for those phones. Select **Yes** to remove the outgoing-access assignments, or select **No** to allow outgoing access.

5. Press **ENTER** or click out of the field to save the change.

ECHO TRUNK NUMBER

For CO Trunk Groups:

When the Echo Trunk Number option is enabled, the system echoes the trunk number, DNIS, DID (or DDI in Europe), and so forth, if the call is being routed out on a trunk. The base digits are combined with the collected trunk number to create an outside number. If the trunk rings into another trunk in the system through an individual trunk, trunk group, or ARS, the trunk number is dialed as the outgoing number.

> **NOTE** | If the collected trunk number is incomplete or invalid, the collected digits are not used for the outside number, to avoid the system from dialing the wrong outside number.

For SIP Peer Trunk Groups:

The Echo Trunk Number is similar to the Echo Trunk Number flag for CO Trunk Groups. This flag is typically used when calls ring into a SIP trunk group from another SIP PBX. When this flag is enabled and a SIP trunk group rings into another trunk in the system through an individual trunk, trunk group, or Automatic Route Selection, the digits dialed from the SIP side are echoed as an outgoing number. This flag is set to No by default.

The following items outline a typical use case for this flag:

- A SIP PBX is configured as a SIP trunk group on the MiVoice Office 250. The SIP PBX does not have local PSTN trunks, but the administrator wishes to have the SIP PBX phones use the MiVoice Office 250's PSTN trunks.

- Set the SIP trunk group's ring-in destination to ARS. Set the SIP trunk group's Echo Trunk Number flag to "Yes".

- When the SIP PBX dials 444-555-1000 and routes the call to the MiVoice Office 250, the MiVoice Office 250 receives the SIP PBX call and then routes the SIP PBX call out ARS dialing 444-555-1000.

To program the Echo Trunk Number:

1. Do one of the following:

    - Select System – Devices and Feature Codes – **CO Trunk Groups**, and then select the trunk group number.

    - System – Devices and Feature Codes – SIP Peers – SIP Trunk Groups – *<SIP trunk group>* – **Trunk Group Configuration**.

2. Select **Echo Trunk Number**.

3. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

4. Press **ENTER** or click out of the field to save the change.

ENABLE HOOKFLASH

You can enable or disable the Hookflash [Recall] feature for each trunk group. If disabled, phone users cannot use the Hookflash feature code (330) while using the trunks in the trunk group. Hookflashes [Recalls] dialed through ARS are ignored by the trunk groups with hookflash [recall] disabled.

To enable hookflash:

1. Select System – Devices and Feature Codes – **CO Trunk Groups**.

2. Select the trunk group number.

3. Select **Enable Hookflash**.

4. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

5. Press **ENTER** or select another field to save the change.

CAMP-ONS ALLOWED

The Camp On feature can be enabled or disabled for each trunk group. If disabled, users placing outgoing calls will hear busy signals when all trunks in the group are in use or unavailable. If enabled, users will be able to camp on and wait for an available trunk. See **page 526** for details about trunk Camp-Ons.

To enable Camp-Ons Allowed:

1. Select System – Devices and Feature Codes – **CO Trunk Groups**.

2. Select the trunk group number.

3. Select **Camp-Ons Allowed**.

4. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

5. Press **ENTER** or select another field to save the change.

ISDN DATA CALLS ALLOWED

This is a BRI-only trunk. This option allows you to prevent ISDN data calls from being routed to trunk groups containing trunks that cannot support data calls, such as loop start trunks.

- When set to **Yes**, the system can route ISDN data calls to this trunk group, and you are allowed to add B-channel trunks only to the group.

- If the flag is set to **No**, the system will not route ISDN data calls to this trunk group, and you can add any type of trunk, except private networking B-channel trunks, to the group. Any non-B-channel trunks must be removed from the CO trunk group before the flag can be set to Yes. If you move multiple trunks into a CO trunk group that has this flag enabled, you will be warned that none of the trunks will be added to the group if any of the trunks to be added are non-B-channel trunks.

To enable ISDN Data Calls Allowed:

1. Select System – Devices and Feature Codes – **CO Trunk Groups**.

2. Select the trunk group number.

3. Select **ISDN Data Calls Allowed**.

4. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

5. Press **ENTER** or select another field to save the change.

DAY AND NIGHT RING-IN TYPES

Ring-in type is determined separately for day and night modes.

To set the Ring-In Type:

1. Do one of the following:
- Select System – Devices and Feature Codes – **CO Trunk Groups**, and then select the trunk group number.
- System – Devices and Feature Codes – SIP Peers – SIP Trunk Groups – <*SIP trunk group*> – **Trunk Group Configuration**.

2. Select **Day** (or **Night**) **Ring-In Type**, and then select the option from the list.

3. Press **ENTER** or click out of the field to save the change. When using an extension list for ring-in or hunt groups, do not exceed 30 phones per list. The system can send ring signal to up to 30 phones simultaneously. The Ring-In Type can be the same for both modes, or any combination of the following can be used:
- **Single**: The trunk group can ring at a single extension number or a hunt group. This can be a local or off-node extension number. When you select Single, you must also enter the destination extension in the Ring-In column. Use one of the following methods:

  Method A

  a. Select the current Value, then enter the new value in the text box.

b. Press **ENTER**. A screen displays what is associated with the number entered.

c. Click **OK**. The new number appears in the field.

Method B

a. Right-click the existing value. An option box appears.

b. Click **Change Extended Value**. The Change Extended Value dialog box appears.

c. Select the appropriate device type, and then click **Next**. The list of devices appears. To view devices in a list only, click **List**.

d. Select the device you want to use, then click **Finish**. The selection appears in the Ring-In field.

• **Multiple**: The trunk group can ring in to a list of phones, extension lists, and/or applications (but not hunt groups). The list can include local or off-node device extension numbers. Set the ring-in destination as described on **page 508**.

• **DISA**: If the trunk group is to be used for DISA, a security code can be assigned by selecting the Ring-In field and entering the code in the text box. To prevent unauthorized access to the public network, *all* trunk groups using DISA should have a security code.

• **Call Routing Table**: If the trunk is used for Caller ID [CLID in Europe], DID/DNIS, or ANI, the calls can be routed according to the information sent by the CO. For more information about Call Routing Tables, see "Call Routing Tables" on **page 248**. When you select Call Routing Table, you must also enter the destination table in the Ring-In column, using one of the following methods:

Method A

a. Select the current Value, then enter the new value in the text box.

b. Press **ENTER**. A screen appears displaying what is associated with the number entered.

c. Click **OK**. The new number appears in the field.

Method B

a. Right-click the existing value. An option box appears.

b. Click **Change Extended Value**. The Change Extended Value dialog box appears.

c. Select **Call Routing Table**, and then click **Next**. The list of tables appears. To view tables in a list only, click **List**.

d. Select the table you want to use, then click **Finish**. The selection appears in the Ring-In field. Loop start trunks that are connected to paging equipment should not have any ring-in designations. They should be reserved for internal use only.

• **Collected Digits**: This ring-in type indicates that the collected DID [DDI in Europe] or DNIS digits (plus the base digits) should be used as the destination extension. This helps to keep the number of call routing table entries down to a minimum when routing calls to extensions.

Both trunk groups and call routing table entries can use the new ring-in type. When the new Ring-In Type of Collected Digits is selected, the Ring-In Destination field is empty.

If the collected digits plus the base digits do not make up a valid ring-in destination, the call is routed to the primary attendant. Valid ring-in destinations include on- or off-node phones, on- or off-node hunt groups, trunk groups, individual trunks, voice mail applications, automated attendants, and ARS.

SEND STATION EXTENSION/USERNAME TO ATTACHED PBX

This flag allows the user name and extension in the ISDN or SIP setup request message is sent on an outgoing ISDN or SIP peer trunk call. All intercom calls that route externally from the system through an ISDN circuit or SIP peer trunk group will send the user name and extension for the caller ID name and number instead of the Calling Party Number or Name information. The default value is set to No. For details, see "Caller ID Forwarding" on **page 534** and "Caller ID, DNIS, and ANI" on **page 172**.

Consider the following when using this flag:

- This flag is enabled on a per trunk group basis.

- ISDN and SIP peer trunks (non-private networking) use this field.

- In the case of SIP, it will influence the "From," "Contact," and "P-Asserted-Identity" fields in the SIP headers.

- The user name and extension that are sent on the outgoing call are always the extension that has originally placed the call. For example, a voice mail application calls an extension that is forwarded to an outside number. The system sends the user name and extension from the voice mail application and not the extension that was forwarded to the outside number. If the voice mail application answers and transfers an incoming call to an extension that is forwarded to an outside number, the system sends the user name and extension from the voice mail application.

- Hunt group calls that are processed by an extension or open architecture interface (OAI) are governed by the same conditions as listed in the tables beginning on **page 536**. If the user name has not been programmed, the system does not include the name information on the outgoing call.

- The user name may not be processed on an outgoing call across nodes if the outgoing PRI is using "display IE" instead of "facility IE" for caller ID name. The user name on the Mitel system is not passed immediately between nodes, which is why the "display IE" does not contain the user name. The "display IE" on an outgoing call is sent immediately where a "facility IE" is delayed.

- All System and Manual Forward types apply to this feature.

- This flag applies to both local and remote node calls.

    - When set to **No**, the system refers to the phone Calling Party Number or Calling Party Name field in System\Devices and Feature Codes\Phones\Local\<*extension*>. When the phone's Calling Party Number or Calling Party Name field is programmed with a number or name, the programmed information is sent. If the field is blank, no caller ID information is sent.

    - This flag has no effect on incoming trunk calls being forwarded or redirected out another trunk.

To propagate a calling phone username and extension to an attached PBX:

1. Do one of the following:

   - Select System – Devices and Feature Codes – **CO Trunk Groups**, and then select the trunk group number.

   - System – Devices and Feature Codes – SIP Peers – SIP Trunk Groups – *<SIP trunk group>* – **Trunk Group Configuration**.

2. Select **Send Station Extension/Username to Attached PBX**.

3. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

4. Press **ENTER** or click another field to save the change.

PROPAGATE ORIGINAL CALLER ID

Propagate Original Caller ID allows the system to pass the caller ID name or number on an outgoing ISDN or SIP peer trunk call if the call has not been answered by the system (extension, voice mail, hunt groups, or OAI application) or for transfer announcement calls. This option is for customers that want to route incoming calls from the MiVoice Office 250 back to the PSTN through ISDN lines or SIP peer trunks. By default this flag is set to No. For details, see "Caller ID Forwarding" on **page 534**, "Caller ID, DNIS, and ANI" on **page 172**, and "Do Not Propagate Original Caller ID to P-Asserted Identity" on **page 524**.

This flag applies to three scenarios: 1) incoming trunk calls forwarded/deflected back to the PSTN through ISDN lines or SIP peer trunks; 2) incoming trunk calls answered by a voice mail Call Routing Announcement (CRA) and transferred back to the PSTN through ISDN lines or SIP peer trunks; 3) incoming trunk calls answered by an operator and transferred back to the PSTN through ISDN lines or SIP peer trunks. Consider the following when using this flag:

- This flag is enabled on a per trunk group basis.

- ISDN and SIP peer trunks (non-private networking) use this field for the outgoing call. Do not mix trunk types in the trunk groups when using this feature.

- This flag must be enabled on the outgoing trunk group for the caller ID information to be processed.

- All Manual Forward types apply to this feature when used to redirect the call to an outside number.

- OAI Deflect applies to this feature.

- If the incoming caller ID information only has a number and no name, the system only sends the number that was received.

- Incoming caller ID information that is blocked on the incoming call is not sent on the outgoing call. If the incoming call is blocked, the system only passes the generic information received for the blocked call. Example of these generic blocked messages are: Private, Out of Area, Restricted, Unassigned Number, or Blocked Caller. The system does not send information when no caller ID information was received.

- The outgoing trunk group must be ISDN or SIP for the Caller ID information to be processed on the outgoing call.

- The incoming call cannot be answered by the system. This flag is ignored If the call is answered and transferred.

- Calls to a hunt group that have not been answered are considered unanswered calls.

- Calls that have been played an announcement through a hunt group announcement are still considered an unanswered call.

- Enabling this flag applies to local node and remote node calls that route through the trunk group that has this flag enabled.

- The public telco may remove the caller ID name that is sent by the system and only pass the caller ID number.

NOTE	The system or network administrator must confirm with the CO as to whether or not the Forward Original Call-ID for Unanswered Calls feature is supported, as the Public Switched Telephone Network (PSTN) may reject this call that is "masquerading caller ID."

To route incoming calls from the MiVoice Office 250 back to the PSTN via ISDN lines or SIP peer trunks:

1. Do one of the following:

   - Select System – Devices and Feature Codes – **CO Trunk Groups**, and then select the trunk group number.

   - System – Devices and Feature Codes – SIP Peers – SIP Trunk Groups – *<SIP trunk group>* – **Trunk Group Configuration**.

2. Select **Propagate Original Caller ID**.

3. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

4. Press **ENTER** or click another field to save the change.

DO NOT PROPAGATE ORIGINAL CALLER ID TO P-ASSERTED IDENTITY

The administrator can set this flag when the Dynamic Extension Express (DEE) feature is used, and a DEE destination is reached via a SIP trunk and the *Propagate Original Caller ID* flag is enabled.

Some SIP providers will not allow (authenticate) calls via their network that contain source numbers that the SIP provider does not recognize in the P-Asserted-Identity header. If certain providers receive call INVITEs with non-recognized numbers, they fail the call since they cannot authenticate the INVITE.

This new flag, when set to **Yes**, removes the original incoming caller ID information from only the P-Asserted-Identity header of the INVITE going out to the SIP trunk. The end result is a call where the SIP INVITE contains the correct data for the SIP provider to know the call is from a valid customer (the call is really originating from this MiVoice Office 250), even though some of the headers will contain the original Caller ID (from the original call that came into the MiVoice Office 250)

CALLING PARTY NAME

You can program the Calling Party Name option for the trunk group. For more information about the Calling Party name option, see "Calling Party Name" on **page 411**.

CALLING PARTY NUMBER

You can program the Calling Party Number option for the trunk group. For more information about the Calling Party name option, see "Calling Party Number" on **page 411**.

FORCE TRUNK GROUP CALLING PARTY NAME AND NUMBER

When selected, the system uses the Calling Party Name and Calling Party Number configured on the trunk group, regardless of any other caller ID configuration. When cleared, the system follows either the Calling Party Name and Calling Party Number configured on the device making the call, or from forwarded caller ID (see Caller ID forwarding). The default setting is No.

NOTE	Some SIP trunk providers require a valid caller ID to be presented as part of authentication. If this is the case, enable this flag and choose a valid Calling Party Number to ensure this is always sent to the SIP peer.

To enable the Force Trunk Group Calling Party Name and Number option:

1. Do one of the following:

   • Select System – Devices and Feature Codes – **CO Trunk Groups**, and then select the trunk group number.

   • System – Devices and Feature Codes – SIP Peers – SIP Trunk Groups – *<SIP trunk group>* – **Trunk Group Configuration**.

2. Select **Force Trunk Group Calling Party Name and Number**.

3. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

4. Press **ENTER** or click another field to save the change.

WAIT FOR ISDN CALLER ID INFORMATION

The Wait for ISDN Caller ID Information timer determines the amount of time the system waits (in seconds) for the incoming ISDN Facility message that contains the caller ID name before routing the call to the ring-in destination. This timer only applies to incoming ISDN calls that use Facility messages instead of Display messages for providing caller ID name.

The default value of this timer is 0 seconds, which is no wait delay. This timer should only be adjusted if the system is receiving Facility messages from the telco where the name is not appearing on the forwarded or redirected calls. Contact your ISDN provider to verify if you are receiving Facility messages for caller ID name. Or, use the diagnostic port on the T1/E1 board and enable "DON ALL" and review the incoming ISDN messages for Facility or Display messages (remember to disable the DON information after reviewing the information by typing DOFF).

Some ISDN central offices use a Facility IE message to transfer a calling party name to the system. This is necessary to allow the PSTN to look-up the calling party's name. The Facility

message containing the caller ID name maybe delayed for several seconds after receiving the original incoming setup request. This timer provides the delay mechanism that allows the incoming call information to be processed so that the caller ID name can be forwarded on the outgoing call. When this timer is set to a value greater than 0, the system waits for the Facility message or the timer to expire before processing the call. When the timer is set to 0, the system processes ISDN calls as it did in versions earlier than v3.0 not waiting for the Facility message with name.

If you want to route incoming calls back to the PSTN, the outgoing call should not be sent until the system has received the "facility IE" message. After the system has the calling party name, the system can then forward the calling party name back to the PSTN. An ideal value for this timer is about 5 seconds.

See **page 534** for details about Caller ID Forwarding.

To set the Wait for ISDN Caller ID Information timer:

1. Select System – Devices and Feature Codes – **CO Trunk Groups**.

2. Select the trunk group number.

3. Select **Wait for ISDN Caller ID Information**.

4. In the **Value** column, enter the time (in seconds).

5. Press **ENTER** or click another field to save the change.

TRUNK CAMP ON AND BUSY TRUNK QUEUE CALLBACK

Camp On and Queue Callback requests allow the caller to wait for an available resource. A phone user waiting for a specific resource (a trunk or a phone) will be served before a phone user waiting for a group (trunk group or hunt group) that contains that specific resource. This includes camped on Emergency Call feature calls. For details about the Emergency Call feature, see "Emergency Calls" on **page 182**.

CAMP ON

When a phone user attempts to select a busy outgoing trunk or trunk group, the system sends a busy signal. The phone user can wait off-hook to camp on until the trunk is available.

A user can camp on to busy resources on other nodes and hears the other node's Music-On-Hold while camped on. When the system has several routing options for a call, it tries to use each route, in order, until it finds a free route. If it is unable to find a free route, the system tries once again to use the first route. If the first route is still busy, the system camps on to the first route.

For programming instructions, see **page 519**. For feature usage instructions, refer to the applicable phone user guide.

BUSY TRUNK QUEUE CALLBACK

If you attempt to select a busy outgoing trunk group and hear a busy signal or if you camp on, you can request to be queued for a callback. When the trunk group is available, the system signals your phone that a trunk in the trunk group is available. Each phone can place only one

queue request at a time. If a second request is made, the first request is canceled and replaced by the second request.

A user cannot queue onto busy resources on another node.

Queue callbacks must be answered before the Queue Callback timer expires. If a callback is not answered, the queue is canceled. If the phone is busy when a trunk in the queued trunk group becomes available, the queue request is placed at the end of the queue list.

For feature usage instructions, refer to the applicable phone user guide.

TOLL RESTRICTION

The following features provide toll restriction on the Mitel system through Class of Service (COS) programming. Classes of Service are described in further detail in the following paragraphs.

- **Trunk Group Toll Restriction**: Designates a trunk group as "subject to toll restriction" or "not subject to toll restriction" in database programming. If subject to toll restriction, phone class of service is checked when the trunk group is selected for placing an outgoing call.

- **Exemption From ARS-Only Restriction**: Designates each trunk group as "exempt from automatic route selection (ARS) only." If exempt, phone users with the ARS-Only class of service can select the trunk group directly.

- **Absorbed Digits**: Allows trunk groups that are subject to toll restriction to ignore, or absorb the first digit(s) dialed. This allows the system to handle the dialed digits just as they would be by the local telephone company or PBX to which the system is connected.

- **Phone and Trunk Group Classes Of Service**: Restricts or allows certain digit patterns when an outside call is dialed. (Trunk group COS is *not* checked when ARS is used.)

- **Toll Strings**: Allow the system administrator to define dialing patterns which are abbreviated to single character wildcards. These programmable wildcards are used frequently in the toll restriction class of service programming and in other system areas such as ARS route group programming, trunk group absorbed digits programming, and others. ***Changing these dialing patterns will affect all of the areas where they are used.***

For programming information, see **page 511** for trunk groups and **page 400** for phones.

TRUNK GROUPS SUBJECT TO TOLL RESTRICTION

Each trunk group can be designated as "subject to toll restriction" or "not subject to toll restriction" in database programming. All trunk groups are subject to toll restriction when the system is in the default state.

- When a phone user selects a trunk group that is designated as "subject to toll restriction," the system checks the database for that phone's class of service, and call cost is determined by the digits that are dialed.

- When a trunk group is chosen that is "not subject to toll restriction," the phone COS is *not* checked, except ARS-Only restriction.

The exceptions to trunk group toll restriction are as follows:

- **Emergency Call Feature**: Uses the designated trunk access code to seize a trunk and dials the programmed emergency number regardless of any phone or trunk group toll restrictions.

- **Automatic Route Selection (ARS)**: Designates all calls using ARS (except Emergency Call feature calls) as always subject to toll restriction, regardless of the selected trunk group's toll restriction programming. Only the phone COS is checked; the trunk group COS is not.

- **Network ARS Calls**: Determines toll restriction for an ARS outgoing call by checking the phone toll restriction based on the database information on the node on which the phone resides, not the node that contains the trunk which the system uses to place the call. The system does not check the trunk COS for ARS calls.

Trunk groups can be programmed as "not subject to toll restriction" to allow phone users to have access to reduced-cost long distance carriers, or to use ring-down lines, dictation machines, voice mail systems, and other auxiliary equipment. When programming unrestricted trunk groups, one of the following five call-cost factors can be selected:

- Free

- Local

- Toll local

- Toll long distance

- Operator/International.

The selected call cost is then used for all calls that are placed using that trunk group.

Whether or not a trunk is marked as "subject to toll restriction," dialing is *not* required to hold the trunk. For example, when a user seizes a line to make an outgoing call—via any method— if there is an incoming call on the line for which ring-in had not yet been detected, the user can talk with the incoming caller without dialing any digits. That is, the call will not be dropped when the Dial Initiation timer expires.

However, even though the system provides protection for "glare" between incoming and outgoing calls, the protection is overridden if the caller is dialing a restricted number. Unless the "Drop Incomplete Outgoing Calls" flag is disabled (see "System Settings" on **page 729**), the call *will* be dropped if the trunk is "subject to toll restriction" and the user dials digits that represent a restricted number for the phone and trunk group being used.

TRUNK GROUP EXEMPTION FROM ARS ONLY

A trunk group may be designated as "exempt from Automatic Route Selection (ARS) Only." This feature allows users with the ARS Only class of service to directly select specified trunk groups by pressing the appropriate trunk group buttons or by entering the appropriate trunk group access codes.

This capability is necessary if trunks are connected to auxiliary equipment, such as analog voice mail systems, dictation equipment, or ring-down lines. When such trunks are designated as exempt from ARS Only, phones with the ARS Only COS (and allowed access) can use the special facilities.

The default assignment for all trunk groups is "not exempt from ARS Only." This means that, by default, phones with the ARS Only COS are denied direct access to the trunk groups.

AREA/OFFICE CODE NUMBERING PLANS AND LOCAL AREA CODES

If any office codes in the local area code have a 0 or 1 as the second digit, the local office codes are probably used as area codes elsewhere, and the system requires special programming to allow toll restriction to work properly. During System-Wide Information database programming, the system administrator can specify the following non-standard numbering plan information:

- **Office Codes Used as Area Codes:** Area codes in other locations are the same as office codes within the system site's area code.

- **Area Codes Used as Office Codes:** Office codes within the system site's area code are the same as area codes assigned to other areas.

- **Local 7- or 10-Digit Dialing**: When this flag is enabled, outgoing calls are identified as having reached the end of dialing if the first digits are not a toll field, equal access field, or a local area code. This function speeds up placement of local seven-digit calls in an area where some local calls require 10 digits.

- **Toll Digit Allowed On Toll Local Calls:** This option applies only if the area and office codes overlap. Callers in the site's area code usually dial a 1 when placing a call within the local area code(s).

- **Toll Digit Required On Toll Long Distance Calls:** This option applies only if the area and office codes overlap. Callers in the site's area code must dial a 1 when placing a call outside of the local area code(s).

The database requires that the system administrator enter a "home" area code for the system site. It also allows up to 16 home area codes for non-toll calling. The system refers to these area codes for toll restriction and call cost.

ABSORBED DIGITS

Trunk groups that are subject to toll restriction can be programmed to "absorb" (ignore) the first digits so that the digits dialed are handled by the system just as they would be by the local telephone company or PBX to which the system is connected. There are two applications for this capability: PBX installations and installations in areas where part of the local office code is absorbed by the central office (CO). Trunk groups that connect to outside trunks in PBX installations are referred to as "PBX trunk groups." Trunk groups that use trunks for which the CO absorbs part of the local office code are referred to as "local trunk groups."

When a Mitel system is installed behind a PBX, users must dial special PBX codes to select the CO trunks. The system trunk group that is used for PBX trunk access must be programmed to absorb PBX digits. The PBX access codes are then programmed as the absorbed digit patterns for that PBX trunk group. Up to 50 absorbed digit patterns are allowed, with a maximum of 48 digits in each.

When a PBX trunk group is used and an absorbed digit string is *not* dialed, the system behaves as though the trunk group is "not subject to toll restriction" and does not perform toll restriction and call cost functions. The call is considered to be an internal PBX call; it is allowed no matter what digits are dialed. The call cost rate is "PBX," and the call is designated as free (000) in the SMDR report.

When a PBX trunk group is used and an absorbed digit string *is* dialed, the system absorbs the PBX access code and behaves as though the trunk group is "subject to toll restriction," using the remaining digits for toll restriction and call cost processing. When the Redial, Speed Dial, or Call Forward features are used, the system automatically inserts a short pause after the absorbed digit string to provide for the delay that occurs while a trunk is seized.

In some rural areas, specific digits (dialed as all or part of the local exchange) are absorbed by the central office, thus reducing the number of digits required to dial local calls. These digits may also be "repeatable." That is, they are absorbed if dialed more than once. To determine if a central office absorbs digits and whether the digits are repeatable, contact the telephone company.

When using this type of trunk, the system must be programmed to recognize the absorbed digits. That is, the absorbed digits must be programmed as an absorbed digit string for any trunk groups that include those trunks. For each local trunk group, only one string is allowed, with a maximum of 48 digits. The only characters allowed in the string are digits (0-9), pound sign (#), parentheses ( ), brackets ([ ]), greater than (>) and less than (<) symbols, and the wildcards.

If the system is not programmed to recognize the absorbed digits for a PBX or local trunk, the system may be left open for users to bypass toll restriction. A user could bypass toll restriction by dialing the digits that are absorbed by the CO before dialing a toll number. If the system does not know that the first digits are absorbed, it will not recognize a toll digit that follows them. However, if the digits absorbed by the CO are programmed as an absorbed digit pattern for the trunk group, the system will recognize them, absorb them, and then recognize any digits that follow as possible toll digits.

When dialed, absorbed digits (PBX or local) appear in the SMDR record if they are not suppressed during SMDR programming. Repeatable (local) absorbed digits will appear in SMDR only once, even if dialed repeatedly.

Absorbed-digits are programmed on a trunk group-by-trunk group basis. In the default state, no trunk group absorbs digits.

EQUAL ACCESS

Under the terms of the final divestiture agreement, Bell Operating Companies provide equal access to all long distance companies. The equal access provision requires each Bell Operating Company to modify existing switching equipment to make it possible for customers to have direct access to all available carriers serving their area code. Prior to equal access, routing long distance calls to long distance companies other than AT&T required a lengthy dialing process usually involving a local access number, authorization number, and public network number.

As central offices are converted to provide equal access, customers served by those offices may choose one of two options for the routing of long distance calls. The options are:

- **Pre-subscription:** Normal "1+" or "0+" dialing procedures may be used and a switching arrangement automatically routes calls to the predesignated long distance company.

- **Non-Pre-subscription:** A long distance company is not predesignated and a five or six digit equal access code identifying a long distance company (10XXX or 101XXXX) is re-

quired to be dialed preceding the desired "1+" or "0+" long distance telephone number to designate the long distance company to handle the call.

Pre-subscription does not limit long distance calling to the predesignated long distance company. An access code (10XXX or 101XXXX) may be dialed to direct a call to the long distance company of choice. Pre-subscription also does not affect the way an operator may be contacted. Any existing non-pre-subscription long distance company operator is called by dialing the carrier's access code followed by "0."

COS

Each phone and each trunk group that is "subject to toll restriction" is assigned toll restriction classes of service (COS) that restrict or allow certain dialing patterns on outside calls. For programming instructions, see "Numbering Plans" on **page 123**.

The classes of service (COS) are programmed individually for phones, Voice Processing applications, and trunk groups. There are separate COS designations for day and night modes. A phone or trunk group can be completely unrestricted or can have any combination of the classes of service.

The first nine COS designations (01–09) have default values. COS designations (10–16) do not have default values and are blank. All of the classes of service are programmable. Each may be specified as an "allowed" or "denied" class of service, dialing patterns can be programmed for each, and each can be assigned to phones and trunk groups as needed. The default values of COS 01-09 are as follows:

- **COS 01 – ARS Only:** *(This is a phone COS only. It is not used for trunk groups. Also, it cannot be used unless the ARS feature is available.)* Calls can only be placed using the Automatic Route Selection (ARS) feature when this restriction is assigned. The user will hear reorder tones when attempting to place a call using any other method. A restricted user can still select individual trunks if the trunks are designated as "exempt from ARS Only," were transferred, were placed on hold, or are recalling or ringing. Trunk restriction determines which trunks in the ARS route group can be selected by the phone or Voice Processing application.

- **COS 02 – Deny Area/Office:** This restriction is divided into eight user groups to allow the use of varying area/office code restriction tables. This is useful for reducing restrictions for some of the phones, Voice Processing applications, or trunk groups while increasing restrictions for others. Each phone, application, and trunk group is assigned a day mode and a night mode user group in database programming. Within each user group, area codes can be designated as restricted, allowed, or extended. Restricting an area code prevents users from placing calls to that area code. Allowing an area code allows all office codes within that area code. Designating an area code as extended allows the system administrator to determine which office codes (up to 800) are allowed or restricted within that area code. For each user group, 800 area codes can be marked as allowed or restricted in the database list, and up to six area codes can be marked as extended.

- **COS 03 – Deny Operator:** Calls to numbers that match the dial patterns for this class of service (defaults to [Q]RN+, [Q]R0, and [Q]RE) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the phone, Voice Processing application, or trunk group being used.

- **COS 04 – Deny Toll Access:** Calls to numbers that match the dial patterns for this class of service (defaults to [Q]TN+ and [Q]TE) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the phone, Voice Processing application, or trunk group being used.

- **COS 05 – Deny International:** Calls to numbers that match the dial patterns for this class of service (defaults to [Q]I+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the phone, Voice Processing application, or trunk group being used.

- **COS 06 – Deny Equal Access:** Calls to numbers that match the dial patterns for this class of service (defaults to Q+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the phone, Voice Processing application, or trunk group being used.

- **COS 07 – Deny Local Calls:** Calls to numbers that match the dial patterns for this class of service (defaults to N+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the phone, Voice Processing application, or trunk group being used.

- **COS 08 – Denied Numbers:** Calls to numbers that match the dial patterns for this class of service (defaults to 1900NXXXXXX+ and 976XXXX+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the phone, Voice Processing application, or trunk group being used.

- **COS 09 – Allowed Numbers:** Calls to numbers that match with the dial patterns for this class of service, defaults to 1(800, 888, 877, 866, 855, 844, 833, and 822)NXXXXXX+, are allowed, even if number also matches a dial pattern in a restricted class of service that is assigned to the phone, Voice Processing application, or trunk group being used.

Calls are only restricted if the dialed patterns match the denied pattern exactly and that is the only class of service you have. Allowed patterns, however, always override denied patterns, even if the numbers are similar. For example, the following table demonstrates what is allowed or denied when the restricted pattern is 976+ and the allowed pattern is 976-1111.

Table 82:   Example of Restricted Dialing Pattern

NUMBER DIALED	ALLOWED OR DENIED?
97 (partial match)	Allowed
9761234 (exact match with "denied" pattern)	Denied
9761111 (exact match with "allowed" pattern)	Allowed

System Speed Dial numbers can be programmed to bypass COS restrictions on a system-wide basis. If the option is not enabled, all System Speed Dial numbers are subject to toll restriction.

Because a condition may exist where a critical or life threatening situation needs to be reported, using the Emergency Call feature code will override all toll restrictions and trunk access programming.

In the event that the classes of service for the phone or Voice Processing application and the trunk group conflict, the decision to restrict or allow a number is based on the following ordered

checks. The ordered checks apply to **all** calls to outside telephone numbers, including calls forwarded to outside telephone numbers.

- *Phone or Voice Processing application COS is checked.* If the digits dialed **exactly** match a pattern included in a "denied" class of service assigned to the phone or Voice Processing application and not with a pattern included in an "allowed" class of service assigned to the phone or Voice Processing application, the number is restricted.

- *If ARS is not being used and forward to an outside number is not being programmed, trunk group COS is checked.* If the digits dialed **exactly** match a pattern included in a "denied" class of service assigned to the trunk group and not with a pattern included in an "allowed" class of service assigned to the trunk group, the number is restricted.

- *If ARS is being used and a forward to an outside number is being programmed, phone or Voice Processing application COS is checked.* If the digits dialed **partially** match a pattern included in a "denied" class of service assigned to the phone or Voice Processing application and not with a pattern included in an "allowed" class of service assigned to the phone or Voice Processing application, the number is restricted.

- *If a forward is being programmed to an outside number, trunk group COS is checked.* If the digits dialed **partially** match with a pattern included in a "denied" class of service assigned to the trunk group and not with a pattern included in an "allowed" class of service assigned to the trunk group, the number is restricted.

- *If none of the above cases apply,* the number is allowed. If the trunk group is "not subject to toll restriction," neither the trunk group nor phone COS is checked, unless the call was placed using ARS. All ARS calls are subject to **phone** toll restriction only.

TOLL STRING DIALING PATTERN WILDCARDS

Toll strings are dialing patterns that are abbreviated to single-character wildcards. These programmable wildcards are used frequently in the toll restriction class of service programming and in other parts of the system, including ARS route group programming, trunk group absorbed digits programming, and others.

| NOTE | Changing these dialing patterns will affect all of the other parts of the system where they are used. |

Toll strings can contain any digit 0–9, #, or *. In addition, a variety of special characters may be entered into the database to reflect particular digit strings, hookflashes, or special digit strings (as described below).

Each of the following toll string wildcards can be reprogrammed and/or renamed to meet the customer's requirements. The programmable toll string wildcards are as follows:

- **Operator Access (R):** Represents the digit string that is required to reach a telephone company operator. In the default state, this dialing pattern is 0.

- **Toll Access (T):** Represents the digit(s) required when using long distance service. In the default state, this toll string is 1.

- **International (I):** Represents the digits required for international dialing. In the default state, the dialing patterns for this toll string are 011 and 01.

- **Equal Access (Q):** Represents the digits required for access to secondary carriers using equal access dialing. In the default state, this toll string is 101XXXX, 10NXX, and 100XX.

- **Any Toll String (S):** Indicates that any of the designated patterns can be dialed at that point in the digit string. In the default state, the dialing patterns for this toll string are International (I), Operator Access (R), and Toll Access (T).

NON-PROGRAMMABLE WILDCARDS

Table 83 shows the special characters that may be used when specifying dialing patterns. These characters are not programmable.

Table 83: Special Characters for Specifying Dialing Patterns

CHARACTER	MEANING
X	Any digit 0–9
A	Any keypad entry (0–9, #, *)
N	Any digit 2–9
Z	Any digit 0 or 1
B	# or *
H	A hookflash [Recall]
E	End of dialing; the pattern will not match if any digits are dialed beyond this point
+	Any additional dialing will be accepted, from this point in the string, with no further checking for a match. This also means that no further dialing is required beyond this point.

Unless otherwise noted, each wildcard character represents exactly one character position in the dialed number. For example, 258999X will accept 258999 plus any digit 0–9.

A character inside brackets indicates an optional pattern within another pattern. For example, the International Access character "I" could be defined as 01[1]. The 01 is followed by an **optional** 1. To add an optional equal access code to the beginning, the pattern would be [Q]01[1].

A range of numbers within parentheses, such as (00–99), indicates a range of digits within a pattern. The digits on either side of the hyphen and all digits that fall within the numerical range are included in the match. The digit strings on either side of the hyphen must be the same length, and the only digits that may appear in the range are 0–9 (#, *, pauses, and hookflashes are not allowed).

Patterns within angle brackets, such as <9>, indicate repeatable patterns within patterns. In other words, no matter how many times the digit string within the brackets is dialed, the system will consider the dialed digits to match the pattern. A repeatable pattern is an entire pattern; no other characters are allowed before or after a repeatable pattern. In other words, a repeatable pattern cannot be included within any other pattern.

CALLER ID FORWARDING

This section contains the following information:

- **Feature Description** below

- Call Processing Forwarding Priorities on **page 541**

- Programming Caller ID Forwarding on **page 545**

- Troubleshooting Caller ID Forwarding on **page 546**

FEATURE DESCRIPTION

Caller ID forwarding allows Call Processing (CP) to send Caller ID information from any incoming services to the public or private network using Primary Rate Interface (PRI) or Basic Rate Interface (BRI) Integrated Services Digital Network (ISDN) circuits, or SIP peer trunks. In the European market Caller ID is known as Calling Line Identification [CLID]. When Caller ID is enabled, the system includes information that identifies the caller to the public or private network.

The Caller ID information can be a phone user name or extension, caller number (calling party number or name), or incoming Caller ID information. The system software can process any incoming caller ID information and then resend this information through an ISDN circuit or SIP peer trunk with some limitations. Caller ID received through an ISDN/analog line/T1 circuit or SIP peer can be processed and sent out to an ISDN circuit or SIP peer. Caller ID information can also be forwarded between nodes.

Caller ID information presented by the system results from any one of the following events:

- A call that is originated from a phone on the system.

- A call that is redirected from an external call.

- The system programmable fields in DB Programming (see **page 545**).

Caller ID for Line Type

Based on the caller ID settings, the following calls are supported with the processing of the Caller ID number or name.

Table 84:   Caller ID Line Type

INCOMING CALL TYPE	OUTGOING CALL TYPE	THE SYSTEM SENDS:
PRI	PRI	Caller ID name and number
	BRI	Caller ID number only[1]
	SIP	Caller ID name and number
BRI	PRI	Caller ID number only[1]
	BRI	Caller ID number only[1]
	SIP	Caller ID number only[1]

Table 84:   Caller ID Line Type (continued)

INCOMING CALL TYPE	OUTGOING CALL TYPE	THE SYSTEM SENDS:
T1 Automatic Number Identification (ANI)	PRI	Caller ID number only[2]
	BRI	Caller ID number only[1]
	SIP	Caller ID number only[1]
T1 Loop Start with Caller ID	PRI	Caller ID name and number
	BRI	Caller ID number only[1]
	SIP	Caller ID name and number
Analog Loop Start with Caller ID	PRI	Caller ID name and number
	BRI	Caller ID number only[1]
	SIP	Caller ID name and number
SIP	PRI	Caller ID name and number
	BRI	Caller ID number only[1]

1. BRI only supports caller ID number and not name. Caller ID name cannot be received or sent on a BRI card.
2. ANI is the same as Caller ID number or Calling Number. If the ANI number is received, this can be processed on an outgoing ISDN circuit.

Incoming calls that have their caller ID information blocked are not sent by the system. Instead, the system passes the generic block name (Private, Restricted, Out-Of-Area, or Blocked). The system can only send caller ID information it receives. If the incoming call does not provide both name and number, the information sent will only include what was received.

NOTES	When the caller ID name is sent to the public telephone network, the name may be removed or replaced by the CO.
	Some SIP trunk providers use the caller ID as part of authentication for outbound calls. If this is the case, calls will fail if the incorrect caller ID is sent to the SIP Peer.

Network Considerations

Keep the following considerations in mind for networked systems:

• An intercom call that routes between nodes then out through an ISDN line that is using a Display message does not send the phone's user name for the caller ID name, but sends the phone's calling party number instead.

• An intercom call that routes between nodes then out through an ISDN that is using a Facility message sends the phone's user name for the caller ID name.

The reason for the user name condition above is for network compatibility for older systems. The user name information between nodes is not sent immediately.

Caller ID Forwarding Scenarios

The following scenarios describe how incoming and outgoing calls are connected through a PRI to another PBX using the Caller ID Forwarding feature.

Scenario #1: An extension or application places an outbound call.

Table 85:   Caller ID Forwarding for Outbound Calls

CALL TYPE	WHEN:	IS SET TO:	THE SYSTEM SENDS:
Outbound External Call	Calling Party Number and/or Name	No entry	No Caller ID information
	Propagate Original Caller ID	--[1]	
	Send Station Extension/Username to Attached PBX	No	
	Propagate Original Caller ID on Transfer	--[1]	
	Calling Party Number and/or Name	an outside telephone number and/or name	Calling Party Number and/or Name
	Propagate Original Caller ID	--[1]	
	Send Station Extension/Username to Attached PBX	No	
	Propagate Original Caller ID on Transfer	--[1]	
	Calling Party Number and/or Name	No entry or an outside telephone number/name[1]	User name and extension from the calling extension
	Propagate Original Caller ID	--[1]	
	Send Station Extension/Username to Attached PBX	Yes	
	Propagate Original Caller ID on Transfer	--[1]	

1. This flag or entry has no affect on the situation because the other flags that are enabled have priority, the call condition does not apply, or caller information is not provided.

Scenario #2: An extension transfers an inbound or outbound call to an outside number or to an extension that is forwarded to an outside number.

Table 86:   Caller ID Forwarding for Non-Voice Mail Transfer Calls

TYPE OF CALL	WHEN:	IS SET TO:	THE SYSTEM SENDS:
External Call Transfer	Calling Party Number and/or Name	No entry	No Caller ID information
	Propagate Original Caller ID	No	
	Send Station Extension/Username to Attached PBX	No	
	Propagate Original Caller ID on Transfer	No	
	Calling Party Number and/or Name	an outside telephone number and/or name	Calling Party Number and/or Name
	Propagate Original Caller ID	--[1]	
	Send Station Extension/Username to Attached PBX	No	
	Propagate Original Caller ID on Transfer	No	
	Calling Party Number and/or Name	No entry or an outside telephone number/name[1]	User name and Extension from the calling extension. Calling Party Number is not sent
	Propagate Original Caller ID	--[1]	
	Send Station Extension/Username to Attached PBX	Yes	
	Propagate Original Caller ID on Transfer	--[1]	
	Calling Party Number and/or Name	No entry or an outside telephone number/name[1]	Original incoming Caller ID information to the external call
	Propagate Original Caller ID	Yes	
	Send Station Extension/Username to Attached PBX	No	
	Propagate Original Caller ID on Transfer	--[1]	

1. This flag or entry has no affect on the situation because the other flags that are enabled have priority, the call condition does not apply, or caller information is not provided.

Scenario #3: A Voice Mail application transfers an inbound call to an extension that is forwarded to an outside number.

Table 87:   Caller ID Forwarding for Voice Mail Transfer Calls

TYPE OF CALL	WHEN:	IS SET TO:	THE SYSTEM SENDS:
Inbound External Call	Calling Party Number and/or Name	No entry or an outside telephone number/name[1]	User name and Extension from the calling extension. Calling Party Number is not sent.
	Propagate Original Caller ID	-- [2]	
	Send Station Extension/Username to Attached PBX	Yes	
	Propagate Original Caller ID on Transfer	--[2]	
	Calling Party Number and/or Name	No entry or an outside telephone number/name[1]	Calling Party Number and/or Name
	Propagate Original Caller ID	Yes	
	Send Station Extension/Username to Attached PBX	No	
	Propagate Original Caller ID on Transfer	No	
	Calling Party Number and/or Name	No entry or an outside telephone number/name[1]	User name and extension from the calling extension
	Propagate Original Caller ID	Yes	
	Send Station Extension/Username to Attached PBX	No	
	Propagate Original Caller ID on Transfer	Yes	

1. This field does not apply to this type of device. Voice Mail applications do not have Calling Party Number field.
2. This flag or entry has no affect on the situation because the other flags that are enabled have priority, the call condition does not apply, or caller information is not provide.

Scenario #4: Inbound call is immediately routed to an outside number using Single Ring-in Destination, Station Forwarding, and OAI Deflect.

Table 88:   Caller ID Forwarding for Incoming Unanswered Calls

TYPE OF CALL	WHEN:	IS SET TO:	THE SYSTEM SENDS:
Inbound External Call	Calling Party Number and/or Name	No entry or an outside telephone number/name[1]	No Caller ID information
	Propagate Original Caller ID	No	
	Send Station Extension/Username to Attached PBX	--[2]	
	Propagate Original Caller ID on Transfer	--[2]	
	Calling Party Number and/or Name	No entry or an outside telephone number/name[2]	Original incoming Caller ID information to the external call
	Propagate Original Caller ID	Yes	
	Send Station Extension/Username to Attached PBX	--[2]	
	Propagate Original Caller ID on Transfer	--[2]	

1. The Calling Party Field will only be sent if the phone places an outbound call or if a phone intercoms another extension that is forwarded to an outside number. Incoming calls routed to a phone that are forwarded or processed by OAI from a phone to an outside number will not provide the Calling Party Number. The Calling Party Number will not be sent if an incoming unanswered call immediately routes to an outside number through Call Routing Table or directly to a trunk group or through phone forward of any type.
2. This flag or entry has no affect on the situation because the other flags that are enabled have priority, the call condition does not apply, or caller information is not provided.

Scenario #5: An extension calls another extension that is routed to an outside number through phone forward, OAI, or Send to Destination. Same Node or remote Node intercom calls are processed the same.

Table 89:   Caller ID Forwarding for Intercom Calls

TYPE OF CALL	WHEN:	IS SET TO:	THE SYSTEM SENDS:
Intercom Call	Calling Party Number and/or Name	No entry	No Caller ID information
	Propagate Original Caller ID	--[1]	
	Send Station Extension/Username to Attached PBX	No	
	Propagate Original Caller ID on Transfer	--[1]	
	Calling Party Number and/or Name	An outside telephone number/ name	See footnote [2]
	Propagate Original Caller ID	--[1]	
	Send Station Extension/Username to Attached PBX	No	
	Propagate Original Caller ID on Transfer	--[1]	
	Calling Party Number and/or Name	No entry or an outside telephone number/ name[1]	User name and extension from the calling extension. Calling Party Number is not sent.
	Propagate Original Caller ID	--[1]	
	Send Station Extension/Username to Attached PBX	Yes	
	Send Station Extension/Username to Attached PBX	--[1]	

1. This flag or entry has no affect on the situation because the other flags that are enabled have priority, the call condition does not apply, or caller information is not provided.
2. The Calling Party Field will only be sent if the phone places an outbound call or if a phone intercoms another extension that is forwarded to an outside number. Incoming calls routed to a phone that are forwarded or processed by OAI from a phone to an outside number will not provide the Calling Party Number. The Calling Party number will not be sent if an incoming unanswered call immediately routes to an outside number through Call Routing Table or directly to a trunk group or through phone forward of any type.

CALL PROCESSING FORWARDING PRIORITIES

The Call Processing forwarding priorities are designed to provide maximum flexibility without burdening the calls that do not require Caller ID.

If the Emergency Calling Party Number is configured, Call Processing uses the Emergency Calling Party Number as the outgoing ISDN Caller ID [CLID]. If the Emergency Calling Party Number is not configured, Call Processing uses the phone Calling Party Number, trunk group Calling Party Number, or phone extension depending on the "Propagate ID on Transfer" and

"Send Station Caller ID to Attend PBX" flags. See **page 522** for details about these fields. See the flowchart on page 543 for phone level Caller ID Forwarding call flow.

If the voice mail Caller ID [CLID] is configured, Call Processing uses the voice mail Caller ID [CLID] when the Message Notification Retrieval application makes an outgoing call. When a Call Routing Application (CRA) makes an outgoing call, Call Processing will forward the Caller ID [CLID] based on the "Propagate Original Caller ID on Transfer" flag (Message Notification/ Retrieval applications only). See **page 523** for details about this field. See the flowchart on page 544 for voice mail level Caller ID Forwarding call flow and the flowchart on page 545 for SIP voice mail level Caller ID Forwarding call flow.

The trunk group Calling Party Number and Calling Party Name are used only if no other Calling Party Number can be determined. Call Processing uses the Calling Party Number and Calling Party Name that is defined in the trunk group only if the original Caller ID [CLID] or phone Caller ID [CLID] information is not set. The trunk group Caller ID [CLID] has the least precedence than other Caller IDs. The flowcharts on pages 543 through 545 show how Call Processing forwards the Caller ID [CLID] based on the configuration. However, if the Force Trunk Group Calling Party Name and Number flag is set on the Trunk Group, then the Calling Party Name and Number specified at the Trunk Group level takes priority over any other Caller ID information.

Station makes an outgoing call

Force trunk group calling party name & number?

No

Emergency number?

Yes → Is emergency calling party number defined?

Yes → Number: Emergency calling party number
Name: Station calling party name if defined

No

Is OAI application changing calling party name and/or number?

Yes → Name: OAI calling party name if defined
Number: OAI calling party number if defined

No

Transferring a trunk call?

Yes → Trunk's propagation original Caller ID

Yes → Name: Holding party name
Number: Holding party number

No

Station's forward original CID?

No

Send station CID to attach PBX?

Are station calling party number and/or name defined?

Yes → Name: Station calling party name
Number: Station calling party number

No

Are trunk group calling party number and/or name defined?

Yes → Name: Trunk group calling party name
Number: Trunk group calling party number

No

Yes → Name: Station Username
Number: Station extension

Name: N/A
Number: N/A

Yes → Are trunk group calling party number and/or name defined?

No

Are station calling party number and/or name defined?

No

Yes

PROGRAMMING CALLER ID FORWARDING

To support Caller ID Forwarding:

Program the following fields under System\Devices and Feature Codes\CO Trunk groups\<*trunk group*>:

• Send Station Extension/Username to Attached PBX on **page 522**

• Propagate Original Caller ID on **page 523**

• Wait for ISDN Caller ID Information on **page 525**

Program the following fields under System\Devices and Feature Codes\Phones\<*phone*>:

• Calling Party Name on **page 411**

• Calling Party Number on **page 411**

If a SIP trunk group is used for routing outbound calls, program the following fields under System\Devices and Feature Codes\SIP Peers\SIP Trunk Groups\<*SIP peer trunk*>\Trunk Group Configuration:

• Send Station Extension/Username to Attached PBX on **page 522**

• Propagate Original Caller ID on **page 523**

TROUBLESHOOTING CALLER ID FORWARDING

The following table lists troubleshooting information for the Caller ID Forwarding feature.

Table 90:   Caller ID Forwarding Support Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
The customer reports incoming trunk caller ID does not forward to an outbound PSTN call.	The original incoming trunk call did not have caller ID.	Enable service with the CO provider to ensure incoming trunk calls have caller ID.
	The system administrator has not enabled the Propagate Original Caller ID flag for the correct CO trunk group or SIP peer trunk group, if the call is being routed through the SIP peer trunk group.	Enable the "Propagate Original Caller ID for Unanswered Calls" flag for the correct CO trunk group.
	The outbound call is sent out a non-ISDN PSTN line.	For Caller ID Forwarding, the customer must use an ISDN line to forward caller ID.
	The original trunk is answered by the system (either through a Voice Mail application, human operator, or Unified Communicator).	Currently, the system does not support propagating caller ID for incoming trunk calls that are answered and then transferred back out to the PSTN.
	If the answer is none of the above, the CO may not support the ISDN message for propagating caller ID.	Call the CO provider to determine why the ISDN caller ID information is not forwarded. If the CO cannot resolve this issue, contact Technical Support and provide the following information:
		1. the ISDN switch type where outbound calls was made; and
		2. freeze of this call scenario.
The customer reports phone caller ID does not forward to external PBX.	The system administrator has not enabled the "Send Station Extension/Username to Attached PBX" flag for the correct CO trunk group or SIP peer trunk group, if the call is being routed through the SIP peer trunk group.	Enable the "Send Station Extension/Username to Attached PBX" flag for the correct CO trunk group.
	The outbound call is sent out a non-ISDN PSTN line.	For Caller ID Forwarding, the customer must use an ISDN line to forward caller ID.
	An unsupported SIP gateway is used to connect the system to an external PBX.	Verify that the SIP gateway has been officially tested. If the SIP gateway is officially supported, call Technical Support and provide the following information:
		1. the ISDN switch type where outbound calls was made,
		2. type of gateway used to connect to external PBX,
		3. ISDN switch type settings on SIP gateway, and
		4. freeze of this call scenario.

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Table 90: Caller ID Forwarding Support Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
The customer reports phone caller ID name (and only name) is not propagating to an external PBX. The customer is using a SIP gateway to communicate to an external PBX.	Two specific conditions cause this to happen: 1. the outbound ISDN line is configured as a DMS-100; and 2. the originating extension is calling out ARS and ARS is routing the call to a remote node.	There is a scenario where phone Caller ID Forwarding is used over a networked ARS configuration, and the phone user name will not be passed in the ISDN calling party name field. If the outgoing ISDN line uses the "display IE" method to transfer the calling party name, the calling party name will not use the phone's user name. Instead of the user name, the system uses the calling party name as the name.

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NODE TRUNK GROUPS

Node trunk groups are created when the T1/PRI, or E1/PRI Switch Type is set to Private Networking or IP Private Networking. At the main level, you can program the description and user name for each node trunk group. You can also see which bay contains the circuits used for this network connection.

VIEWING NODE TRUNK GROUP TRUNK LISTS

When the T1/PRI or E1/PRI module Switch Type is programmed for Private Networking, all of its B-Channels are automatically assigned to a node trunk group. You can double-click Trunks to view this list, but you cannot change it.

To view a Node Trunk Group List:

1. Select System – Devices and Feature Codes – **Node Trunk Groups**.

2. Select the trunk group number.

VIEWING OR CHANGING NODE TRUNK GROUP INFORMATION

Node trunk group trunks show the following information:

- **Description and User name**: All node trunk groups should have a description and a user name.

- **Associated Bay Number**: This is the bay number where the module associated with this node trunk group is located. This field is for reference only and cannot be changed.

When the T1/E1/PRI or Dual T1/E1/PRI module's Switch Type is programmed for Private Networking, all of its B-Channels are automatically assigned to a node trunk group. You can view the list but you cannot change it.

To program the Description or Username fields:

1. Select System – Devices and Feature Codes – **Node Trunk Groups**.

2. Select the trunk group number.

3. Select the **Description** or **Username** field, and then type the new name in the box. The description that appears in all node trunk group lists in the database can be up to 20 characters long. The username, that appears on display phones, can have up to 10 characters.

4. Press **ENTER** or click another field to save the change.

PROGRAMMING NODE TRUNK GROUP OPTIONS

You can program the following Node Trunk Group options:

- Emergency Outgoing Access on **page 549**

- Day or Night Outgoing Access on **page 550**

- Search Algorithm on **page 550**

- Camp-Ons Allowed on **page 551**

- ISDN Data Calls Allowed on **page 551**

EMERGENCY OUTGOING ACCESS

There are separate lists for phones with emergency outgoing access in day and night modes. By default, the automatic phone list (Auto: All Phones) is assigned to Day/Night Emergency Outgoing Access.

To add phones that will have emergency outgoing access for the node trunk group:

1. Select System – Devices and Feature Codes – **Node Trunk Groups**.

2. Select the trunk group number.

3. Select **Emergency Outgoing Access**, and then select either **Day Mode** or **Night Mode**.

4. Right-click anywhere in the right side of the window. An option box appears.

5. Select **Add To List**. A window appears prompting for the device type to include.

6. Select the device types (you can use the SHIFT or CTRL key to select more than one item), and then click **Next**. The items with details appear. To view items in a list only, click **List**.

7. Select the appropriate items, then select **Add Items**. When you have added all the desired devices, click **Finish**. The selections appear in the list. To view programming options, double-click the extension number.

To delete items or destinations from the list:

1. Select **Emergency Outgoing Access**, then select either **Day Mode** or **Night Mode**.

2. Select the item, right-click, and then select **Remove Selected Items**.

WARNING

Responsibility for Regulatory Compliance.

It is the responsibility of the organization and person(s) performing the installation and maintenance of Mitel Advanced Communications Platforms to know and comply with all regulations required for ensuring Emergency Outgoing Access at the location of both the main system and any remote communication phones. Remote IP and SIP phones may require gateway access to nearby emergency responders.

Emergency Call phone numbers include:

- 911, the default for Mitel systems located in the U.S.
- 999, the default for Mitel systems located in the European market and used primarily in the United Kingdom U.K.
- If applicable, 112, an emergency number used widely in Europe outside of the U.K.
- 112, the default for Mitel systems located in Australia.
- Any emergency number, such as for a police or fire station, that is appropriate for the location of the main system and/or remote phones.

DAY OR NIGHT OUTGOING ACCESS

There are separate lists for phones with outgoing access in day and night modes. By default, the automatic phone list (Auto: All Phones) is assigned to Day/Night Outgoing Access when a node trunk group is created.

To add phones that will have outgoing access for the node trunk group:

1. Select System – Devices and Feature Codes – **Node Trunk Groups**.

2. Select the trunk group number.

3. Select **Outgoing Access.**

4. Select either **Day Mode** or **Night Mode**.

5. Right-click anywhere in the right pane. An option box appears.

6. Select **Add To List**. A window appears prompting for the device type to include.

7. Select the device types (you can use the SHIFT or CTRL key to select more than one item), and then click **Next**. The items with details appear. To view items in a list only, click **List**.

8. Select the appropriate items, then select **Add Items**. When you have added all the desired devices, click **Finish**. The selections appear in the list. To view programming options, double-click the device.

To delete destinations from the list:

1. Select System – Devices and Feature Codes – **Node Trunk Groups**.

2. Select the trunk group number.

3. Select **Outgoing Access.**

4. Select either **Day Mode** or **Night Mode**.

5. Select the destinations, right-click, and then select **Remove Selected Items**.

SEARCH ALGORITHM

See "Search Algorithm" on for a feature description.

To change the trunk group searching algorithm:

1. Select System – Devices and Feature Codes – **Node Trunk Groups**.

2. Select the trunk group number.

3. Select **Search Algorithm**.

4. In the **Value** column, select the setting.

5. Press **ENTER** or click outside the field to save the change.

CAMP-ONS ALLOWED

The Camp On feature can be enabled or disabled for each node trunk group. If disabled, users placing outgoing calls will hear busy signals when all trunks in the group are in use or unavailable. If enabled, users will be able to camp on and wait for an available trunk.

To enable or disable the Camp On Allowed option:

1. Select System – Devices and Feature Codes – **Node Trunk Groups**.

2. Select the trunk group number.

3. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

4. Press **ENTER** or click out of the field to save the change.

ISDN DATA CALLS ALLOWED

This is a BRI-only trunk. When set to Yes, it allows you to add only B-channel trunks to the trunk group. With the flag set to No, you may add any types of trunk, except private networking B-channel trunks. Any non-B-channel trunks must be removed from the node trunk group before the flag can be set to Yes. If you move multiple trunks into a node trunk group that has this flag enabled, you will be warned that none of the trunks will be added to the group if any of the trunks to be added are non-B-channel trunks. By default, the flag is set to No.

To enable or disable the ISDN Data Calls Allowed option:

1. Select System – Devices and Feature Codes – **Node Trunk Groups**.

2. Select the trunk group number.

3. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

4. Press **ENTER** or click out of the field to save the change.

HUNT GROUPS

This section contains the following information:

- **Feature Description** below

- Programming ACD Hunt Groups on **page 569**

- Programming Local Hunt Groups on **page 571**

- Programming Local Hunt Group Options on **page 572**

- Node-Spanning Hunt Groups on **page 587**

- Programming Remote (Off-Node) Hunt Groups on **page 587**

FEATURE DESCRIPTION

The Hunt Group feature permits calls to be placed to a group of phones and to be automatically transferred to an available phone in the group. You can program up to 300 hunt groups. Hunt group lists can contain individual phones or extension lists. Non-ACD hunt group phones must reside on the same node; off-node devices must be ACD hunt group members.

The order in which hunt group phones receive incoming calls is determined by the search type selected (see **page 585**). A phone or extension list (**page 500**) can appear in a single hunt group more than once, and it can appear in multiple hunt group lists, if desired.

Hunt groups have their own extension numbers (defaults to 2000–2299). Individual phones within the hunt group can be called using their assigned extension numbers.

Two additional hunt group types are available:

- UCD Hunt Groups (**page 555**), which is included with the system license

- ACD Hunt Groups (**page 555**), which is a software license feature

For feature usage instructions, refer to the applicable phone user guide.

HUNT GROUP CALL DISTRIBUTION

When an intercom or outside call is transferred or rings in to the pilot number, it circulates through the hunt group in linear or distributed order until answered, as described below.

- **Linear order**: Incoming calls always start circulating by ringing at the first phone (or extension list) on the hunt group list that is stored in the database.

```
LINEAR HUNT GROUP CALL PROCESSING
                              Hunt group pilot
                              number 2000 called
                                      |
                                      v
Calls will always        HUNT GROUP MEMBERS:
begin search here  --->   EXT. 1000
                          EXT. 1001
                          EXT. 1002
                          EXT. 1003
```
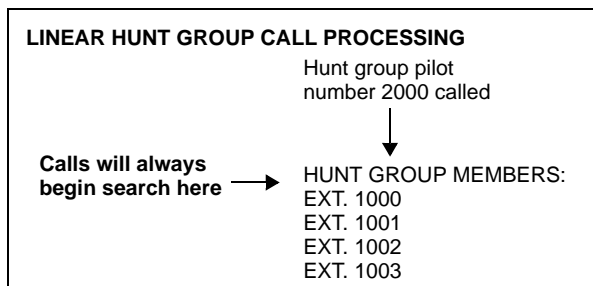
- **Distributed order**: To even out the call load, distributed order shifts the starting point of each call. When a phone user receives a call, the next phone or extension list in the hunt group receives the next incoming call. If the receiving phone is busy (or all phones in the extension list are busy), or if there is no answer before the No

  Answer Advance timer expires, the call goes to the next phone or extension list on the hunt group list.

```
DISTRIBUTED HUNT GROUP CALL PROCESSING

                                Hunt group pilot
                                number 2000 called

                                         |
                                         v
  1st call begins here  ───▶  EXT. 1000
  2nd call begins here  ───▶  EXT. 1001
  3rd call begins here  ───▶  EXT. 1002
  4th call begins here  ───▶  EXT. 1003
```
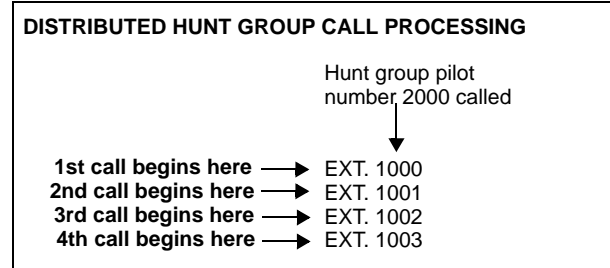
> **NOTE** | For information about the distribution types available in ACD hunt groups, see **page 556**.

If an extension list is included in a hunt group, a call will ring all phones on the extension list when it reaches that point in the hunt group list. Therefore, to create an "all ring" type of hunt group, you can program the hunt group as either linear or distributed and then assign an extension list as the only hunt group member.

A system-wide flag titled "Single Idle Time for All Hunt Groups" invokes an Automatic Call Distribution (ACD) algorithm that sends an incoming call to the agent station with the longest idle time in all the HG queues to which that station belongs. For agent stations belonging to multiple HGs, this feature allows calls to be distributed to other stations having the longest idle time, regardless of the station idle time status in an individual HG.

HUNT GROUP CALL PROCESSING

Hunt group phones receive the following indications when a call is ringing in:

- If an outside call is ringing, the phone designated to receive the call first shows ring flash on the associated individual trunk button (if it has one) *or* a CALL button until the call is answered or the No Answer Advance timer expires and the call moves to the next phone.

- If hunt group camp on is allowed and all phones on the hunt group list are unavailable (busy, in Do-Not-Disturb, or with hunt group remove enabled), an intercom or outside call will camp on, and the phones will receive the following indications:

  - *Busy phones:* Receive hunt group camp-on tones (if enabled) and display. If there is an individual trunk button associated with the trunk, it shows ring flash. A programmable phone flag can be used to disable the camp-on tones for hunt group calls. For programming instructions, see "Camp-Ons Allowed" on **page 551**.

  - *Phones with Hunt Group Remove enabled:* Receive camp-on tones and display. If there is an individual trunk button associated with the trunk, it shows ring flash.

  - *Phones with Do-Not-Disturb enabled:* Receive no camp-on indications, but if there is an individual trunk button associated with the trunk, it shows ring flash.

  - *The first phone that becomes available:* Receives ringing and a flashing trunk or Call button. The Camp On and ring flash indications end at the other phones. (The associated trunk button will be steadily lit at those phones to show that the trunk is busy.)

- If hunt group Camp On is *not* allowed and all phones on the hunt group list are unavailable, the hunt group members will not have any Camp On indications. Callers will hear one of the following:

    - *Intercom callers* will hear busy signals.

    - *Callers on a trunk that does not have to be seized for the system to return busy signal* (such as a T1 E&M) will hear busy signals.

    - *Callers on a trunk that has to be seized for the system to return busy signal* (such as a non-T1 Loop Start) will hear ringing, and the call will be sent to the primary attendant.

    - *ISDN callers* will hear busy signals.

Phones within the hunt group can receive direct trunk ring-in, intercom, forwarded, or transferred calls to their individual extension numbers without affecting other phones in the hunt group.

A flag titled "Return Automatic Call Distribution (ACD) Calls to Hunt Groups" can be set for individual Hunt Groups (HG). When enabled, this feature re-queues a call to the front of the Camp On queue for the HG from which the call came. It allows a calling party to immediately return to the front of the HG queue if the assigned agent station goes into Do-Not-Disturb (DND) mode. If the flag is not enabled, the calling party will continue to ring until the No Answer Advance timer expires or the agent removes DND from the station and answers the call.

Trunk groups can be programmed to ring in directly to either a pilot number or extension number(s). If assigned to a pilot number, ring in for the trunk group cannot be assigned to any other extension number(s).

If an outside call rings in to a pilot number that does not have hunt group phone assignments, the call is sent to the primary attendant. If a phone user attempts to transfer a call to an invalid pilot number, the call is placed on individual hold, and the phone user hears reorder tones. A phone user attempting to place an intercom call to an invalid pilot number will hear reorder tones, and the display will show NO MEMBERS IN HUNT GROUP.

Hunt group programming affects the Call Forwarding feature in the following ways:

- **Hunt group calls follow unconditional forward:** If a phone location in a hunt group is in the unconditional call forward mode to another phone, calls to the hunt group will follow the phone forwarding request. A forwarded hunt group call will ring at the forwarding destination until the No Answer Advance timer expires. If it is not answered before the timer expires, the call will return to the hunt group and continue circulating through the hunt group list.

- **Conditional forward dependent on timer interaction:** Phones that are busy and have their calls forwarded conditionally (no answer, busy, or unavailable) to another phone will receive the hunt group call (if the phone is not busy) until the Forward No Answer timer expires. At this point, if the hunt group's No Answer Advance timer has not expired, then the call will be forwarded. Once the No Answer Advance timer expires, the call will circulate to the next phone in the hunt group list.

- **Hunt group calls will not forward to some destinations**: Hunt group calls will not forward to voice mail, outside numbers, or system forwarding paths.

- **Announcement and overflow stations can forward hunt group calls**: If an announcement or overflow station has call forward enabled, hunt group calls will follow the forward, and the forwarding destination phone will act as the announcement or overflow station.

- **Hunt groups can receive forwarded calls**: Phones can forward calls to a hunt group's pilot number.

Hunt groups can be assigned as message centers and/or alternate message sources for individual phones.

HUNT GROUP TIMERS

See "Timers" on **page 576** for details.

RECALL DESTINATION PHONES

An incoming call to a hunt group immediately registers the Recall timer if there is a recall destination identified for that hunt group. If a call is not answered by a hunt group phone before this timer expires, the call will exit the hunt group and begin ringing at the recall destination phone. The call will remain at this destination until it is answered or the caller hangs up.

If there is no recall destination phone, the call will remain in the hunt group until it is answered or the caller hangs up.

If the recall destination is not available or not reachable, the call will remain in the hunt group until it is answered or the caller hangs up.

HUNT GROUP REMOVE/REPLACE AND DO-NOT-DISTURB

Hunt group members can temporarily stop hunt group calls from ringing at their phones by entering the Hunt Group Remove feature code as described below. If a phone is assigned to more than one hunt group, this halts calls from all hunt groups. Hunt Group assignments cannot be removed individually. DND can also be used to halt Hunt Group and other calls to the phone.

| NOTE | This feature has no effect on ACD hunt groups. See **page 558** for ACD login/logout information. |

When the Hunt Group Remove feature is enabled, the user will still receive the camp-on display and tone, and the individual trunk button flashes, if one exists, for calls to the hunt group. The phone continues to receive calls placed to its extension number. Hunt group overflow and announcement stations cannot block hunt group calls using this feature.

If a phone user has programmed a button for entering the Hunt Group Remove/Replace feature code, and if that button has a lamp, the lamp will be lit whenever the phone is removed from the hunt group.

UCD HUNT GROUPS

Uniform Call Distribution (UCD) hunt groups provide additional features to improve hunt group efficiency. The features added when UCD is enabled are as follows:

- Announcement and overflow stations pick up unanswered calls when the hunt group phones are busy.

- Hunt group priority ranking will place calls to one hunt group before another at phones that are members of more than one hunt group.

- Hunt group supervisors can monitor outside calls of any member of the hunt group.

NOTE	UCD Hunt Groups is a standard feature on the MiVoice Office 250.

Some phones may be members of more than one UCD hunt group. For this reason, hunt groups are assigned a "priority level." The priority level determines which hunt group's calls should be received first when calls ring in or camp on to several hunt groups at once.

UCD priority levels range from 0 (low) to 75 (high). If a phone is a member of multiple hunt groups that have the same priority level, calls received by those hunt groups will be queued in the order they were received by the system. Non-UCD hunt groups are treated as if they have a priority level of 0.

PRIORITY LEVELS

Calls ring in or camp on to all hunt groups

PRIORITY LEVEL

1	HUNT GROUP Ext. 2000
8	HUNT GROUP Ext. 2001
0	HUNT GROUP Ext. 2002

ORDER CALLS ARE RECEIVED

2
1
3
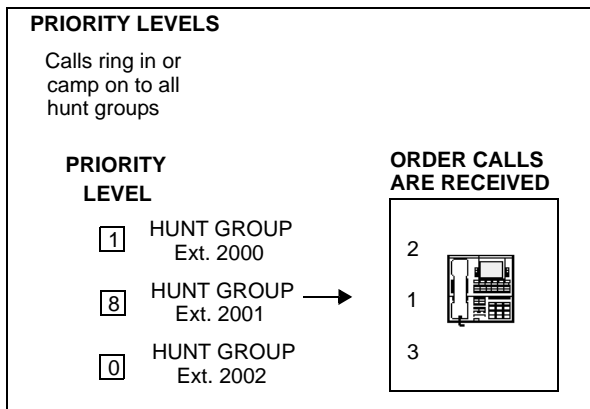
ACD HUNT GROUPS

Automatic Call Distribution (ACD) can be programmed to distribute hunt group calls to equalize call time or call count among the available members. Using the System OAI Events feature, ACD hunt groups can be programmed to send call information records to an external device connected to the system, such as Contact Center Suite.

NOTE	This feature requires the ACD Hunt Group software license.

ACD hunt groups can use the standard hunt group features described in "ACD Hunt Groups" on , and/or the UCD features described on "UCD Hunt Groups" on , if the UCD Hunt Group feature is enabled.

ACD Hunt Group Call Distribution

When an intercom or outside call is transferred or rings in to the ACD hunt group, it can circulate in linear or distributed order, as described on "Hunt Group Call Distribution" on or using one of the following ACD distribution methods:

- **Longest Idle**: Sends an incoming call to the phone that has not been involved in a call to this hunt group for the longest period of time. (It does not count calls that were received through other hunt groups, direct ring-ins, or transfers.

LONGEST IDLE DISTRIBUTION

ACD HUNT GROUP
Ext. 2000

	TIME IDLE
Ext. 1000	(3 min.)
Next call will be sent here → Ext. 1001	(10 min.)
Ext. 1002	(6 min.)
Ext. 1003	(1 min.)
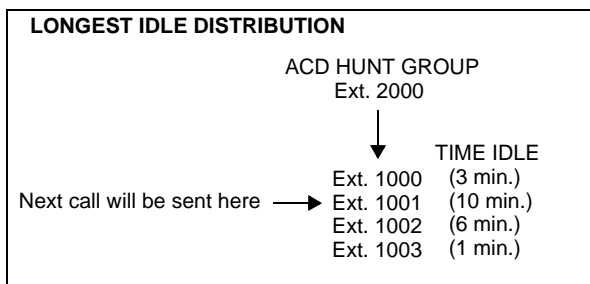
NOTE	The Restart ACD Idle Time Upon Login flag affects how the longest idle value is calculated. For more information, see "Restart ACD Idle Time Upon Login" on **page 584**.

- **Balanced Call Count**: Sends each incoming call to the phone that has received the fewest calls through this hunt group, balancing the call load. (It does not count calls that were received through other hunt groups, direct ring-ins, or transfers.)

```
BALANCED CALL COUNT DISTRIBUTION
                    ACD HUNT GROUP
                       Ext. 2000

                        ▼ NUMBER OF CALLS
                   Ext. 1000  (10 calls)
                   Ext. 1001  (16 calls)
                   Ext. 1002  (6 calls)
Next call will be sent here ──▶ Ext. 1003  (3 calls)
```
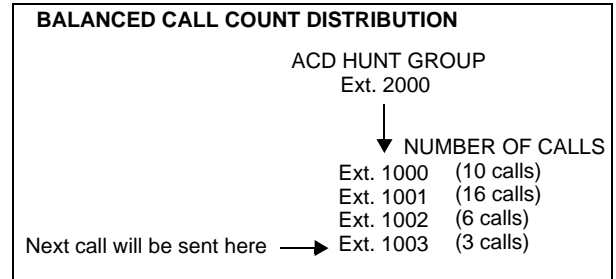
When an agent logs in to a balanced call count ACD hunt group or comes out of DND mode, the system checks the average call count for that ACD hunt group and assigns that average count to the phone logging in. This prevents the agent who is logging in from receiving all of the incoming calls until the call count of the lowest agent catches up with his call count. For example, if there are 5 agents logged in whose call counts are 5, 5, 5, 10, and 10, the average call count of 7 will be assigned to the next agent who logs in.

If an extension list is included in an ACD hunt group set for Longest Idle or Balanced Call Count distribution, it will treat each phone in the extension list as a separate agent; it will not ring all of the phones on the list at once. If the hunt group is set for linear or distributed order, a call will ring at all phones on an extension list at once when the call reaches that point in the hunt group list.

ACD Agent ID

ACD hunt group members are referred to as "agents." Agents log in to the ACD hunt group to receive calls and log out to halt ACD hunt group calls. An ACD hunt group can be programmed to circulate calls to agents in two ways:

- **Agent IDs**: Assigns each agent an Agent ID number which he or she enters during the login procedure (described below). The hunt group calls are routed to logged in agents, according to their Agent ID number instead of their phone extension. Because the Agent ID is not associated with any phone extension, the agent can use any phone in the system to log in and does not have to use the same phone every time.

- **Members**: Sends calls to the phones where agents are logged in, using a list of phones instead of IDs.

> **NOTICE**
>
> For optimum system performance, no more than 1000 Agent IDs should exist in any one hunt group, and no more than 2000 Agent ID entries should exist in all hunt groups combined.

ACD Agent IDs can be included in Extension Lists which allow several ACD Agents to receive a call at once.

When a call camps on to an ACD hunt group that uses Agent IDs, only the agents currently logged in to the hunt group receive camp-on indications.

ACD hunt group supervisors receive visual camp-on displays if they are programmed as members of the hunt group and have the ACD Agent Logout feature enabled.

ACD Agent Login/Logout

ACD hunt group members are referred to as "agents." Agents can log in to and out of the ACD hunt group at any time. While logged in, the agent will receive calls through the ACD hunt group. When the agent is logged out, calls to that ACD hunt group will bypass the phone. (The Hunt Group Remove/Replace feature, described on **page 555**, does not have any effect on ACD hunt group calls when entered by an ACD agent. However, placing the agent's phone in Do-Not-Disturb mode will halt all ACD and non-ACD hunt group calls.)

Three feature codes can be used for logging in to or out of ACD hunt groups:

AGENT ID IN UCD/ACD HUNT GROUP

HUNT GROUP
Ext. 2000

HUNT GROUP

ANNOUNCEMENT PHONE

Ext. 1222

OVERFLOW PHONE

ID 2370

Agent ID can be logged in or out

RECALL PHONE

- ACD Agent Login

- ACD Agent Logout

- ACD Agent Login/Logout

The first two perform only one operation. The third (Login/Logout) is a toggle feature code that logs the phone in or out of *all* associated ACD hunt groups at once. If the toggle feature code is programmed in a user-programmable button with a lamp, the lamp is lit when the agent is logged in to an ACD hunt group and available to receive calls. The lamp is unlit when the agent is logged out. For feature usage instructions, refer to the applicable phone user guide.

ACD Agent Wrap-Up Timer

Each time an agent ends an ACD hunt group call, the phone goes into "wrap-up" mode, and the Wrap-Up timer is started. Until that timer expires, the agent will not receive another call through any ACD hunt group. (However, the agent can receive non-ACD hunt group calls, direct ring-in calls, and transfers.)

The range of the Wrap-Up timer is 1 to 65,535 seconds. The default value is 15 seconds. It is programmed individually for each ACD hunt group.

If an agent wants to terminate the wrap-up mode before the timer expires, he or she can use the following procedure.

For feature usage instructions, refer to the applicable phone user guide.

Wrap-Up Mode for Holding ACD Calls

There is a system flag called "Wrap-Up Mode For Holding ACD Calls." If enabled, it places an ACD agent's phone in wrap-up mode when an ACD call is placed on hold. (However, the ACD Wrap-Up Duration timer is not activated.) This prevents the agent from receiving additional ACD hunt group calls after he or she places an ACD call on hold while the phone is idle. (The phone can still receive non-ACD calls, as usual.) If the flag is disabled, the agent will be available
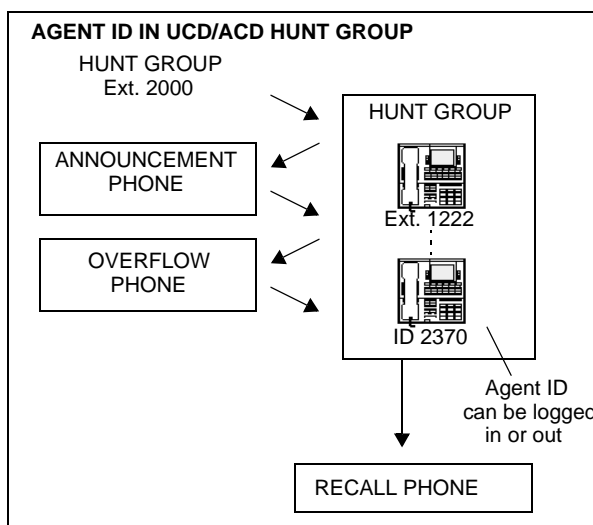
to receive additional ACD calls as soon as an ACD call is placed on hold. In the default state, this flag is disabled.

This flag applies to any type of hold including individual, system, transfer, and conference-wait hold. If the call is terminated or if it is reverse transferred by another phone, the ACD agent's phone will be made available to receive incoming ACD calls.

For feature usage instructions, refer to the applicable phone user guide.

Automatic Connect Flags

When the **Auto Connect** flag is enabled for an agent ID, and the agent is using a headset, ACD hunt group calls will automatically be connected following a short ring burst. (This feature will not work if the agent is not using a headset.)

When the ACD agent logs in, however, the first call rings until the ACD agent answers it. For subsequent calls, the agent hears the ring burst in the headset, and the call is automatically connected. The Auto Connect flag overrides the phone's Transfer-To-Connect Allowed phone flag. It is disabled by default.

When the ACD agent removes the phone from DND, the call may or may not ring until the agent answers it. This is dependent on the **Allow Immediate ACD Auto Connect after DND** flag (under System\Flags). If this flag is enabled, the agent is automatically connected to all calls, including the first one received after exiting DND. If this flag is disabled, the first call rings until the agent answers it, but subsequent calls are automatically connected. By default, this flag is disabled.

REMOTE ACD HUNT GROUPS

The Remote Automatic Call Distribution Hunt Groups software license feature allows ACD hunt groups to span nodes. Node-spanning ACD hunt groups can have either members or ACD Agent IDs.

- **Members**: ACD hunt group members may include off-node phones, off-node single lines, and Hunt Group Member extension lists.

- **ACD Agent IDs**: Unlike members, there are no off-node Agent IDs. Agent IDs are considered global throughout all nodes in which the Agent exists. This means that if you create Agent ID 100 on one node, you must also create Agent ID 100 on all other nodes that have a phone that a user may want to login to the hunt group using that ID.

The Remote Automatic Call Distribution Hunt Groups feature must be available on the node where the hunt group resides as well as any node that wants to have users log in to the node-spanning hunt group. This feature also requires the ACD Hunt Groups software license feature. Because of this fact, several cases must be considered. The following table shows the different displays and results for logging-in to and out of ACD hunt groups with the different combinations of the feature and the ACD Hunt Group enabled/disabled.

Table 91:   Log In Displays and Results for Remote ACD Hunt Groups

STATION NODE ENABLED	HUNT GROUP NODE ENABLED	LOG IN DISPLAY	LOG IN RESULT	LOG IN "ALL" DISPLAY	LOG IN "ALL" RESULT
Yes	Yes	AGENT LOGGED INTO HUNT GROUP XXXX	Agent logged-in to hunt group.	AGENT LOGGED INTO ALL ACDS	Agent logged-in to all local and remote groups.
Yes	No	CANNOT ACCESS RESERVED FEATURE	Agent not logged-in to group.	AGENT LOGGED INTO ALL ACDS	Agent logged-in to all local hunt groups, but not remote where feature disabled.
No	Yes	CANNOT ACCESS RESERVED FEATURE	Agent not logged-in to group.	AGENT LOGGED INTO ALL ACDS	Agent logged-in to all local but no remote groups.
No	No	CANNOT ACCESS RESERVED FEATURE	Agent not logged-in to group.	AGENT LOGGED INTO ALL ACDS	Agent logged-in to all local but no remote groups.

Remote ACD Hunt Group Programming

ACD hunt groups have options allowing the addition of phone off-node devices, single line off-node devices, and hunt group member extension lists containing local or off-node phones or both.

The hunt group feature "Remote ACD Software License Feature," part no. 840.0233, must reside the Software License Features folder in database programming. If this feature is not enabled, the off-node options will not appear in the list when programming members for ACD hunt groups.

Operational Changes

The Station Monitor feature is limited to a single node. Members who are logged in to a remote node or who are logged in to the supervisor's node from a remote node cannot be monitored.

Hunt groups with remote members rely on network links between nodes. Node availability and software version compatibility affect hunt group log-in procedures as follows:

• Hunt group members attempting to log in to all hunt groups will be logged in to only those hunt groups on nodes that are reachable. The display on the user device will only indicate log in results for reachable nodes.

• Members logging out of all hunt groups will be logged out of groups on reachable nodes immediately and will automatically be logged out of groups on unreachable nodes as soon as the link to the node is restored. The phone will display MESSAGE PENDING in this case.

The ACD Wrap-Up functions perform as in previous versions with the exception of priority. Due to timing issues involved in sending messages across the network, the following prioritization method is used for hunt groups with remote members:

• The first criterion is priority level. The hunt group with the highest priority gets the available agent first.

- In the event of a tie in priority, hunt groups located locally take priority over those located on remote nodes.

- In cases where ties in priority occur within the local node, the hunt group with the longest camped on call will receive the available agent.

- Given off-node ties in priority, the first hunt group to camp on will receive the available agent.

Maintenance of Node-Spanning Hunt Groups

- With node-spanning hunt groups combined with the requirement that agents must be defined globally, deleting an agent from a hunt group can be done from the node on which the hunt group resides or from a remote node.

  - If an agent is deleted on the node containing hunt group(s) of which the agent is a member, then the agent is removed from all hunt group(s) on that node and is logged out.

  - If the agent is deleted from a remote node, the agent is logged off from that node, but remains a member of any hunt group(s) located on other nodes.

- When the hunt group is deleted, all remotely logged-in phones are logged out.

- If the extension number of an ACD Hunt Group is changed, any local or remote phones that are logged in will reflect the change. If links are down at the time of the change, remote nodes will not receive the change and an import/export process will be necessary to update the information on the affected nodes.

- Removing an Agent ID from a hunt group agent list logs out any phone, local or remote, that is logged in to that hunt group using that Agent ID.

- With the new node-spanning features of ACD hunt groups, changing Agent IDs can be done from two possible locations: on the node where the group resides or on a node where the logged-in phone resides.

  - When changing an Agent ID on the node where the hunt group resides, the remotely logged-in phone using that Agent ID will be logged out even though the Agent ID still exists on the remote node. The hunt group will then contain the new Agent ID in its removed list, making the new Agent ID available for log-in use.

  - When changing an Agent ID on the node on which the logged-in phone resides (a node remote from the hunt group), the remote member will be logged out and the old Agent ID will no longer exist on that node. The hunt group on the original node will contain the old Agent ID in its removed list, because this ID still exists on that node.

- During a database restore on a node containing ACD hunt groups that use Agents IDs, all Agents IDs are placed into each hunt group's removed list. All phones will be logged out during the restore.

- When the extension number of a logged-in phone is changed, the new extension will be logged in with the same Agent ID and will not be disconnected. This applies to phones logged in both remotely and locally.

ANALOG VOICE MAIL HUNT GROUP

A hunt group can be designated as a Voice Mail Hunt Group to enable it to contain the multiple ports of an analog voice mail unit. Each voice mail unit port is programmed in the distribution list like a regular hunt group phone. With this feature, incoming calls to the voice mail unit can be sent to a single pilot number where they can be processed even if one port is busy or out of service.

| **NOTE** | This feature requires the Voice Processor Analog Hunt Groups software license. |

If the Voice Mail Hunt Group is assigned as the message center for a phone, it is called after the Message Wait timer expires. When the voice mail unit answers the call, the called party's mailbox number is automatically dialed.

| **NOTE** | Analog voice mail units should use the Silent Message feature code (367) instead of the Message feature code (365). |

If the voice mail hunt group is a system forwarding point. For details, see "System Speed Dial" on **page 470**. Or, if a phone is forwarded to the Voice Mail Hunt Group, the mailbox number that is dialed when the voice mail unit answers is the original destination phone's extension number. For example, when the principal phone is called, the call is sent to the first system forwarding point, and then to a voice mail forwarding point where the principal phone's extension number is dialed.

The Voice Mail Hunt Group can also be assigned as the alternate message source for each of the voice mail ports. Then, when a port leaves a message at a phone, and the user responds to the message, the hunt group will be called instead of the individual port. For more information, "Alternate Message Source" on **page 311**.

UCD AND ACD HUNT GROUP FEATURES

The following features are available to UCD or ACD hunt groups.

Announcement and Overflow Stations

Announcement and overflow stations are usually Voice Processor Applications that receive unanswered calls when all of the hunt group phones are unavailable. The same Application can be used for the Announcement and the Overflow, or a different number can be used for each. Also, each of the hunt groups can have the same Announcement and Overflow Application, or they can be assigned different Applications. In a Mitel system network, the

Announcement and Overflow Applications can be off-node devices. For more information, refer to "Appendix A: Private Networking," in the *MiVoice Office 250 Installation Manual* .
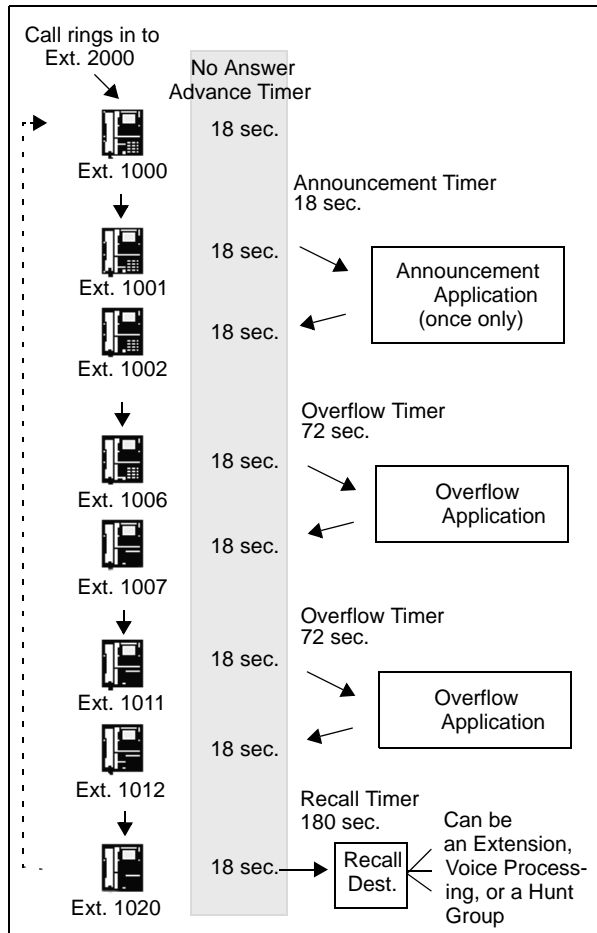
Announcement Stations: When a call rings in or is transferred to a hunt group, the Announcement and Recall timers start. The incoming call circulates through the hunt group until it is answered or the Announcement timer expires. When this timer expires, the unanswered call is picked up by the designated announcement station, which is a playback device that answers the call and plays a message. Meanwhile, the call continues circulating through the hunt group (unless it was sent to a Voice Processing application and then transferred to a phone). If it is answered by an available hunt group phone while the announcement phone is connected to the call, the call will leave the announcement phone. Each direct ring-in call will transfer to the announcement station only once. If it remains unanswered when the Overflow timer expires, the call will be sent to the overflow station. Or, if there is no overflow station, the call will be sent to the recall destination phone when the Recall timer expires.

Overflow Stations: The Overflow timer is started when the Announcement timer expires or, if there is no announcement station, when the call is received by the hunt group. If an incoming hunt group call is unanswered when the Overflow timer expires, the call is picked up by an overflow station. The overflow station is a playback device that answers the call and plays a message. Meanwhile, the call continues circulating through the hunt group (unless it was sent to a Voice Processing application and then transferred to a phone). If the call is answered by an available hunt group phone while the overflow station phone is connected to the call, the call will leave the overflow phone. The Overflow timer restarts each time the unanswered call

leaves the overflow station. An unanswered call will return to the overflow station each time the Overflow timer expires until the call is answered at a hunt group phone or it is sent to the recall destination phone.

The following table shows the path that an incoming hunt group call follows for all possible combinations of announcement and overflow stations:

Table 92:   Announcement and Overflow Station Call Paths

ANNOUNCEMENT STATION	OVERFLOW STATION	CALL PATH
No	No	Call remains in hunt group.
Yes	No	Call goes to announcement station only once, after the Announcement timer expires.
No	Yes	Call goes to the overflow station after each expiration of the Overflow timer.

If an announcement or overflow station has call forward enabled, hunt group calls will follow the forward, and the forwarding destination phone will act as the announcement or overflow station.

Voice Processing Applications as Announcement or Overflow Stations

When a Voice Processing application is used as an announcement or overflow station, calls to the application automatically stop circulating through the hunt group if the caller selects a valid digit translation option (such as pressing a digit to select an extension from a Call Routing Announcement menu), and the call is transferred. This allows the application to send the call to other phones without the call being "pulled back" into the hunt group when a hunt group phone answers the call. However, if the caller does not dial a valid digit translation option, the call will be pulled back if a hunt group member answers.

A Call Routing Announcement application's message can be programmed to include the caller's queue position and/or estimated wait time. The queue position announcement tells the caller how many calls are ahead of his or her call. This includes calls being served and waiting calls (however, all calls being served count as one call). The estimated wait time is based on a programmed Average Connect Time Per Call multiplied by the number of calls ahead of the caller in the queue, divided by the number of available hunt group members (*average connect time per call $\times$ number of waiting calls $\div$ available members*). For more information, see "UCD Hunt Groups" on **page 555**.

If a Voice Processing application is used as an announcement or overflow destination or as the recall destination, and the system is unable to communicate with the Voice Processor, outside calls will not be sent to the announcement or overflow destination. They will continue to camp on to the hunt group.

Agent Help

The Agent Help feature allows a phone user to request help from a designated "Agent Help Extension" during a 2-party or 3-party call. When the request-for-help call rings, the Agent Help Extension can choose to join the call or reject the request. The Agent Help Extension can be a supervisor or other phone, an extension list, or a hunt group.

For feature usage instructions, refer to the applicable phone user guide.

| NOTE | This feature requires the Agent Help software license. |

In database programming, there are two fields for each phone that affect how this feature operates:

- **Agent Help Extension**: Determines the phone that is called when the Agent Help feature code is entered at the phone. This can be set to any valid extension number for a phone, extension list, or hunt group, or it can be set to "None."

- **User-Keyed Extension**: Allows the user to select the phone that will receive the Agent Help request, even if a default Agent Help Extension is programmed as described above.

Table 93: Agent Help Options

AGENT HELP EXTENSION	USER-KEYED	RESULT
Ext. number	Disabled	The Agent Help Extension is called when the feature code is entered.
Ext. number	Enabled	The user is prompted to enter the desired number after the feature code is entered. If a number is not entered before the dialing timer expires, the programmed Agent Help Extension number is dialed automatically.
None	Enabled	The user is prompted to enter the desired number after the feature code is entered.
None	Disabled	The user hears reorder tone if the feature code is entered.

When a phone user enters the Agent Help feature code, a private call is placed to the phone's Agent Help Extension. While the call is ringing at the supervisor's extension, neither the requesting user nor any other parties on the call can hear the private call ringing. If the supervisor answers the call, the system creates a conference to include the supervisor in the requesting user's original call.

If the Agent Help Extension is a phone, the microphone is muted, and the supervisor cannot be heard unless he or she presses the MUTE button. If the Agent Help Extension is a single-line phone, the supervisor *can* be heard as soon as the conference is established. In either case, the supervisor can hear all other parties on the call.

In a network setting, the Agent Help Extension does not need to reside on the same node as the user requesting Agent Help.

The type of phone determines how the Agent Help Extension is alerted:

- *Display Phone:* If the Agent Help Extension is a display phone, the display shows that the incoming private call is an Agent Help request and identifies the username of the requesting phone. The Agent Help Extension can choose to reject the call by not answering it or by entering the Agent Help Reject feature code (376). The call appears under the IC (or a CALL button if there is no IC button).

- *Non-display Phones and Single Line Phones:* The Agent Help Request private call rings just as any private intercom call would; there is no way to distinguish it.

If the Agent Help Extension is an extension list or hunt group, the private call circulates as usual until it is answered. If an Agent Help request is not answered before the Forward No Answer timer expires, the request is considered rejected.

To signal to the other parties that the Agent Help Extension has joined the call, a system-wide Agent Help Tone flag can be enabled during database programming. If the flag is disabled, there will be no alerting tone. A timer, called the Agent Help Tone Interval timer, determines how often this tone is generated. If the timer is set to 0, the tone is generated only when the Agent Help Extension enters the call, a party is added to the call, or the call is placed on hold and retrieved.

If the Agent Help feature code is assigned to a phone feature button with a lamp, the lamp status shows the following:

- The lamp flashes when the Agent Help Extension is being called.

- The lamp goes off if the Agent Help request is rejected or the feature is terminated.

- The lamp is lit solidly when the Agent Help Extension is in the conference.

When the requesting phone user hangs up, all parties are automatically disconnected. The Agent Help Extension can leave the call at any time, without affecting the other parties, by hanging up. If the other party (or parties) hangs up first, the requesting phone and the Agent Help Extension remain connected in an intercom call. The requesting phone can cancel the request (or remove the Agent Help Extension from the call) by re-entering the Agent Help feature code, thereby terminating the feature.

If necessary, the requesting phone in an Agent Help call can use the Hold (see the following notice), Transfer, Record-A-Call, or other features during the call, while the Agent Help Extension is connected. However, if any inside party has enhanced speakerphones enabled, the enhanced mode will be disabled when the Agent Help conference begins and must be re-enabled if still desired.

NOTICE

Placing a Record-A-Call call on Hold terminates the RAC feature.

The Agent Help feature will not function in the following cases:

- If the feature is not available, any user attempting to enter the Agent Help feature code will hear reorder tones and, if at a display phone, see a RESERVED FEATURE display.

- The Agent Help Extension cannot be in Do-Not-Disturb mode or have call forwarding enabled. If so, the requesting user will hear reorder tones after entering the Agent Help feature code or entering the desired extension number. The Agent Help request will not follow the forward or any programmed system forwarding.

- If the phone user who enters the Agent Help feature code is on a four-party conference call, the system will send reorder tones to signal that the Agent Help Extension cannot be added. There can be a maximum of four parties in a conference; attempting to add the Agent Help Extension would exceed the maximum.

- If there are no conference circuits available when the phone user enters the Agent Help feature code, the user will hear reorder tones and must try again later when circuits are available.

If the call is being monitored by a hunt group supervisor, the monitoring feature is terminated when the Agent Help Extension joins the call; an Agent Help conference call cannot be monitored.

Hunt Group Supervisors and Station Monitoring

The Station Monitor feature allows hunt group supervisors to monitor the calls of anyone in a specified hunt group.

For feature usage instructions, refer to the applicable phone user guide.

NOTE	Consider the following concepts when using the hunt group feature: • As a courtesy, hunt group members should be notified in advance that their calls may be monitored. In addition, a programmable option can be enabled that sends a tone to the phone being monitored whenever the hunt group supervisor joins an ongoing call. • Call monitoring may be considered illegal in some areas. The end user is solely responsible for knowing whether the use of this feature complies with local law.

In database programming, each hunt group can have one or more phones assigned as the hunt group supervisor(s). The hunt group supervisors can be off node devices. An extension list can be assigned as the supervisor to provide multiple supervisors. The supervisor is usually not a member of the hunt group. If the supervisor is a member of the hunt group, the Hunt Group Remove/Replace feature can be used at any time without affecting the Station Monitor ability. If desired, one phone can be assigned as the supervisor for more than one hunt group.

To monitor a hunt group member's call, the supervisor enters the Station Monitor feature code (321) and dials an extension number. The supervisor is then connected to the call and can hear both parties, but cannot be heard by either one. If the monitored call is terminated, transferred, or placed on hold by the hunt group member, the monitor function is terminated.

In the associated hunt group, the supervisor may monitor any active intercom or CO-to-intercom call (both hunting and non-hunting), including incoming, outgoing, and DISA-to-intercom calls. Conference calls and calls that do not involve hunt group members cannot be monitored.

If the supervisor attempts to monitor a phone that is not on an active call that allows monitoring, the system sends reorder tones, and the supervisor must enter the feature code again to try another number. If the supervisor attempts to monitor a phone that is not in the hunt group or an idle phone in the hunt group, the system sends reorder tones and cancels the Station Call Monitor feature.

The Station Monitor feature requires conferencing resources. If resources are not available when a supervisor attempts to monitor a phone, the supervisor's display will show NO CNF CIRCUITS AVAILABLE, and the monitor will not be allowed. Multiple supervisors can monitor the same phone, providing that a conference resource is available for each supervisor. Conferencing can now be enabled to support up to eight-party conference calls and 20 total conference resources, which allows for a system maximum of two eight-party and one four-party conference. So a maximum of 6 supervisors can monitor 1 extension at a time, providing resources are available.

The supervisor phone cannot use the Agent Help feature while monitoring a call.

Hunt Group Supervisors and Barge-In

The Barge-In feature adds functionality to the existing Station Monitoring feature. When silent monitoring a call, the hunt group supervisors have the following options:

- **Barge-In:** Allows hunt group supervisors to barge-in on a call to help the hunt group member/agent (default feature code 386).

- **Record:** Allows hunt group supervisors to record the call to review it later (default feature code 385). The supervisor may also hang up and continue to record the call until one of the parties on the call hangs up or puts the call on hold. The record option allows the supervisor to record several calls at once and have them delivered to his or her mailbox.

- **Steal:** Allows hunt group supervisors to steal (take away) the call from the hunt group member/agent (default feature code 387).

- **Join and Record:** *(For 6-line display phones only.)* Allows hunt group supervisors to join and record the call simultaneously. This feature is useful if the supervisor wants to review the call later.

Supervisors must be monitoring a call before they can access any of the Barge-In features (barge-in, steal, record, or join and record). Like the Station Monitor feature, Barge-In requires conference circuits. If resources are not available when a supervisor attempts one of these features, the system sends reorder tones and the supervisor will see NO CNF CIRCUITS AVAILABLE on his or her display.

A monitor/Barge-In call is rejected if:

- A hunt group member/agent is on a conference call.

- The hunt group member/agent is on a peer-to-peer (P2P) call.

- A hunt group member/agent is getting agent help.

- Another supervisor has already joined the call.

When the above situations occur, the supervisor sees SILENT MONITOR REJECTED on his or her display.

The Station Monitor and Barge-In features are supported on remote nodes. A supervisor may monitor a hunt-group member/agent as long as he or she is designated as a supervisor of the group that the member/agent is logged into.

There are two system flags (see "System Flags" on **page 751**) and a system timer (see "Barge-In Notification Tone Frequency Timer" on **page 756**) associated with the Barge-In feature in Mitel DB Programming. These programmable fields allow you to control the Barge-In notification tone and display.

For feature usage instructions, refer to the applicable phone user guide.

Group Call Pick-Up

When the Group Call Pick-Up feature is enabled, a call ringing in to a hunt group or one of its phones can be picked up at any other phone. Users can enter the Reverse Transfer feature code (4) and dial a hunt group's pilot number to pick up a call that is ringing in to the hunt group's

pilot number or to any phone within that hunt group. See "Transfer – Reverse Transfer" on **page 339** for details about Reverse Transfer.

The system routes calls for pick-up in accordance with the following priority list. Always beginning with the first phone on the list, the system follows the hunt group list to check each phone in the hunt group and then the overflow station for one type of call at a time. If there is more than one call of the same type at the selected phone, the call that was received by the phone first is picked up. Holding calls and queue callbacks cannot be picked up.

For feature usage instructions, refer to the applicable phone user guide.

Priority of calls routed by the system to a hunt group:

1. Calls to the hunt group pilot number

2. CO calls

    a. Direct ring-in calls

    b. Transferred calls

    c. Recalls

    d. Camped on calls

3. IC calls

    a. Direct ring-in calls

    b. Transferred calls

    c. Recalls

    d. Camped on calls

NOTE	Group Call Pick-up can only retrieve calls from phones that are logged in to the hunt group at the time. You cannot use this feature to pick up calls from members who have logged out using the Hunt Group Remove feature code. Also, Group Call Pick-up cannot be used on ACD Hunt Groups that use Agent IDs. It can only be used on hunt groups that use lists of extensions. If the ACD Hunt Group flag is enabled, the Group Call Pick-up flag is dimmed.

PROGRAMMING ACD HUNT GROUPS

ACD hunt group members are referred to as "agents." Agents log in to the ACD hunt group to receive calls and log out to halt ACD hunt group calls.

NOTE	This feature requires an ACD Hunt Groups software license.

An ACD hunt group can be programmed to circulate calls to agents in the following two ways:

- **Agent IDs**: If the hunt group is programmed to use ACD Agent IDs, each agent is assigned an Agent ID number which the agent enters during the login procedure. The hunt group calls are routed to logged in agents, according to their Agent ID number instead of their phone extension. Because the Agent ID is not associated with any phone extension, the agent can use any phone in the system to log in and does not have to use the same phone every time.

- **Members**: If the hunt group *is not* programmed to use Agent IDs, it will have a list of phones and will send calls to the phones where agents are logged in.

When a call camps on to an ACD hunt group that uses Agent IDs, only the agents currently logged in to the hunt group will receive camp-on indications. ACD hunt group supervisors will receive visual camp-on displays if they are programmed as members of the hunt group and have the ACD Agent Logout feature turned on. ACD Agent IDs can be included in Extension Lists, which allow several ACD Agents to receive a call at once.

VIEWING AGENT ID LISTS

To view an Agent ID list:

Select System – Hunt Group Related Information – **ACD Agent IDs**. ACD Agent IDs are shown in the right pane.

CREATING ACD AGENT IDS

> **NOTICE**
>
> For optimum system performance, no more than 1000 Agent IDs should exist in any one hunt group, and no more than 2000 Agent ID entries should exist in all hunt groups combined.

To create ACD Agent IDs:

1. Select System – Hunt Group Related Information – **ACD Agent IDs**.

2. Right-click anywhere in the right pane. An option dialog box appears.

3. Click **Create Agent ID**. The Create ACD Agent dialog box appears.

4. In the **Starting Extension** list, select the starting extension number (or lowest extension number, if creating multiple IDs).

5. In the Number of Extensions list, enter the number of IDs you want to create, and then click **OK** to continue. The new ACD Agent IDs appear in the list without a description or username.

6. Program the following options:

   - **Description**: The ACD Agent ID description can be up to 20 characters long. To program the description:

     a. Select the current description, then type the new entry in the text box.

     b. Press **ENTER** or click out of the field to save the change.

   - **Auto Connect**: When this flag is turned on, and headset mode is enabled, ACD hunt group calls automatically connect following a single tone. When the ACD agent logs in, however, the first call rings until the ACD agent answers it. For subsequent calls, the agent hears the ring burst in the headset, and the call is automatically connected. When the ACD agent removes the phone from DND, the call may or may not ring until the agent answers it. This is dependent on the **Allow Immediate ACD Auto Connect after DND** flag (under System\Flags). If this flag is turned on, the agent is automatically connected to all calls, including the first one received after exiting DND. If this flag is turned off, the first call rings until the agent answers it, but subsequent calls are auto-

matically connected. The Auto Connect flag overrides the Transfer-To-Connect Allowed phone flag (see **page 417**) and is turned off by default.

To enable the flag:

a. Select the current value, and then select the check box. The field changes to **Yes**. To disable the flag, clear the check box.

b. Press **ENTER** or click out of the field to save your changes.

DELETING ACD AGENT IDS

To delete ACD Agent IDs:

1. Select System – Hunt Group Related Information – **ACD Agent IDs**.

2. Select the agent IDs. You can use the SHIFT and CTRL keys to select more than one item.

3. Right-click, and then click **Delete**.

PROGRAMMING LOCAL HUNT GROUPS

When you double-click the Local node, a list of the existing local hunt groups, if any, appears. You can create or delete hunt groups and/or enter descriptions and usernames. You can then program the hunt group, as described below.

CREATING HUNT GROUPS

To create a hunt group:

1. Select System – Devices and Feature Codes – **Hunt Groups**.

2. Right-click anywhere in the right pane, and then select **Create Hunt Group**. The Create Hunt Group Extension dialog box appears.

3. Enter the starting extension number and the number of extensions, and then click **OK**. The new hunt group appears in the list without a description or username.

4. Select the **Description** field, and then type the new information in the text box. Descriptions can contain up to 20 characters and hunt group usernames can contain up to 10 characters. Do not use slash (/), backslash (\), vertical slash (|), or tilde (~) characters in usernames. Do not use Control characters in descriptions or usernames.

5. Press **ENTER** or click out of the field to save the change.

DELETING HUNT GROUPS

To delete a hunt group:

1. Select System – Devices and Feature Codes – **Hunt Groups**.

2. Select the hunt group extension number.

3. Right-click, and then click **Delete**. The hunt group is automatically removed from the list.

CHANGING HUNT GROUP EXTENSIONS NUMBERS

You can change hunt group extensions, descriptions, and user names.

To change the hunt group extension:

1. Select System – Devices and Feature Codes – **Hunt Groups**.

2. Select the hunt group extension number.

3. Enter the new number in the text box (or scroll to an available number).

To change several hunt group extension numbers at once:

1. Select System Devices and Feature Codes – **Hunt Groups**.

2. Select the hunt groups that you want to change.

3. Right-click, and then select **Batch Extension Change**. The Create Hunt Group Extension dialog box appears.

4. Select the number that you want to assign to the first selected hunt group (the other selected hunt groups will be numbered consecutively after this number).

5. Click **OK**. the hunt groups are automatically renumbered and re-sorted in the list.

PROGRAMMING LOCAL HUNT GROUP OPTIONS

You can program the following local hunt group options, as shown in:

- Agents on **page 573**

- Members on **page 574**

- Supervisors on **page 576**

- Timers on **page 576**

- ACD Agent No Answer – DND Message Additional Text on **page 577**

- ACD Agent No Answer – DND Message Number on **page 578**

- ACD Hunt Group on **page 578**

- Analog Voice Mail Hunt Group on **page 579**

- Announcement Stations on **page 579**

- Audio for Calls Camped onto this Device on **page 580**

- Audio for Calls Ringing this Device on **page 581**

- Audio for Camped-On Announcement Calls on **page 581**

- Camp-Ons Allowed on **page 582**

- Group Call Pick-Up on **page 582**

- Overflow Stations on **page 579**

- Priority Level on **page 583**

- Recall Destination Phone on **page 583**

- Restart ACD Idle Time Upon Login on **page 584**

- Return ACD Calls to Hunt Group on **page 584**

- Search Type on **page 585**

- Send Camp On Notifications to Members in DND on **page 585**

- Use ACD Agent IDs on **page 586**
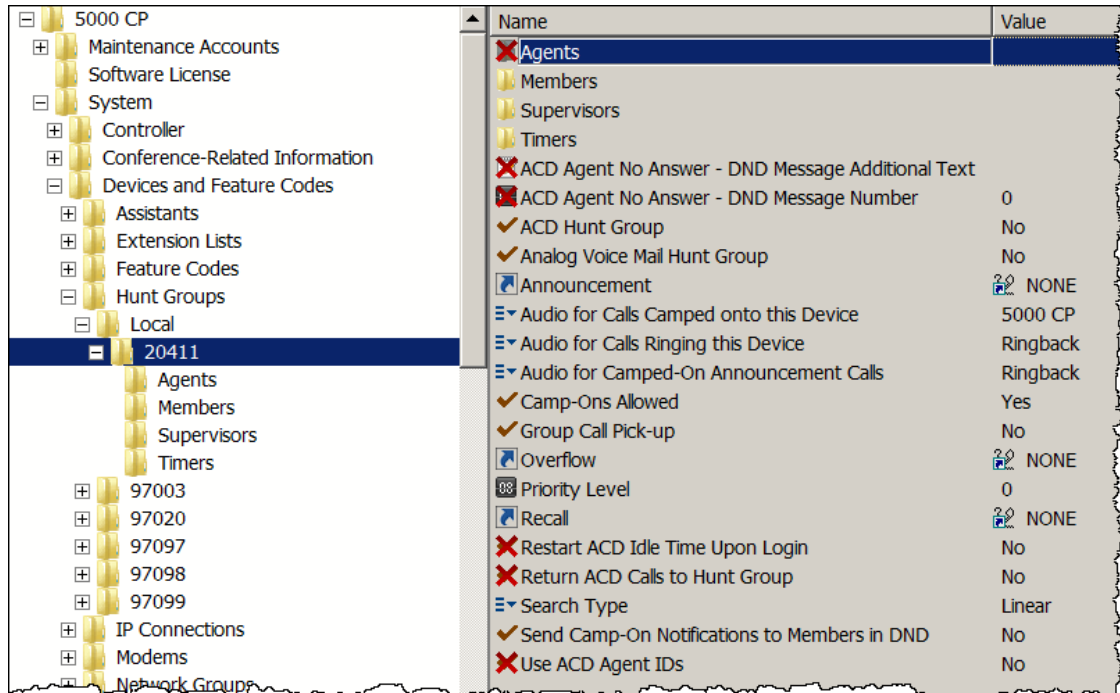
AGENTS

If the hunt group is programmed to use ACD Agent IDs, each agent enters an assigned Agent ID number during the login procedure. The hunt group calls are routed to logged in agents according to their Agent ID number instead of their phone extension. Because the Agent ID is not associated with any phone extension, the agent can use any phone in the system to log in and does not have to use the same phone every time. ACD Agent IDs can be included in Extension Lists (see **page 500**), which allows several ACD Agents to receive a call at once.

To add Agent IDs to the hunt group:

> **NOTICE**
>
> For optimum system performance, no more than 1000 Agent IDs should exist in any one hunt group, and no more than 2000 Agent ID entries should exist in all hunt groups combined.

1. Select System – Devices and Feature Codes – **Hunt Groups**.

2. Select the hunt group extension number.

3. Select **Agents**.

4. Select the extension number.

5. Double-click **Agent IDs** to see the current list of IDs, if any.

> | NOTE | The Agents List is an ordered list, so you must place the agents in the list in the order you want them to be accessed when the hunt group receives a call. |

6. Do one of the following:

   - *To add to the bottom of the list:* Do not select any existing agents, right-click anywhere in the right side of the window, and then select **Add To Agents List**.

   - *To add to the list above an existing agent:* Select the agent, right-click the selected agent, and then select **Add To Agents List**.

   A window appears prompting for the device type to include.

7. Select **ACD Agent ID** and/or **ACD Agent ID Extension List** (you can use the SHIFT or CTRL key to select more than one item), and then click **Next**. The items with details appear. To view items in a list only, click **List**.

8. Select the appropriate items, then select **Add Items**. When you have added all the Agent IDs, click **Finish**. The selections appear in the list. ACD Agent IDs are programmed in Hunt Group-Related Information. To view programming options, double-click the Agent ID number.

To move an agent to another location in the list:

1. Select System – Devices and Feature Codes – **Hunt Groups**.

2. Select the hunt group extension number.

3. Select **Agents**.

4. Double-click **Agent IDs** to see the current list of IDs, if any.

5. Drag and drop the agent to the new position. Or, select the agent to move and press **CTRL** + the up/down arrow to move the agent up or down in the list.

To delete Agent IDs:

1. Select System – Devices and Feature Codes – **Hunt Groups**.

2. Select the hunt group extension number.

3. Select **Agents**.

4. Double-click **Agent IDs** to see the current list of IDs, if any.

5. Select the agent ID(s), and then press **DELETE** on the keyboard, or right-click and select **Remove Selected Items**.

MEMBERS

Prepare a list of the phones and extension lists to be included in each of the hunt groups. If desired, a phone or extension list can appear more than once in a hunt group list or can be in more than one hunt group. If an extension list is included in an ACD hunt group set for Longest Idle or Balanced Call Count distribution, it will treat each phone in the extension list as a separate agent; it will not ring all of the phones on the list at once. If the hunt group is set for linear or distributed order, a call will ring at all phones on an extension list at once when the call reaches

that point in the hunt group list. Therefore, to create an "all ring" type of hunt group, you can program the hunt group as either linear or distributed and then assign an extension list as the only hunt group member.

To add hunt group members:

1. Select System Devices and Feature Codes – **Hunt Groups**.

2. Select the hunt group extension number.

3. Double-click **Members** to see the current list.

NOTE	The Agents List is ordered, so you must place the devices in the list in the order you want them to be accessed when the hunt group receives a call.

4. *To add to the bottom of the list:* Do not select any existing members. Right-click anywhere in the right side of the window, and then select **Add To Members List**.

    To add to the list above an existing member: Select the member, right-click the selected member, and then select **Add To Members List**.

    A window appears prompting for the device type to include.

NOTE	Adding a large extension list can result in slowing down system performance. Adding an Extension List that contains more than 60 members may cause a system slowdown because when the list is called, ALL members of the list are called at the same time.
	When using an extension list for ring-in or hunt groups, do not exceed 30 phones per list. The system can send ring signal to up to 30 phones (60 phones with a PS-1).
	If you exceed the number of phones and the system suffers a slowdown, reduce the number of extension list members.

5. Select the device types (you can use the SHIFT or CTRL key to select more than one item), and then click **Next**. The items with details appear. To view items in a list only, click **List**.

6. Select the items, and then select **Add Items**. When you have added all the members, click **Finish**. The selections appear in the list. To view programming options, double-click the extension number.

To move a member to another location in the list:

1. Select System – Devices and Feature Codes – **Hunt Groups**.

2. Select the hunt group extension number.

3. Double-click **Members** to see the current list.

4. Drag and drop the member to the new position. Or, select the member to move and press **CTRL** + the up/down arrow to move the member up or down in the list.

To delete members:

1. Select System – Devices and Feature Codes – **Hunt Groups**.

2. Select the hunt group extension number.

3. Double-click **Members** to see the current list.

4. Select the members that you want to remove, and then press **DELETE**, or right-click and select **Remove Selected Items**.

SUPERVISORS

(*Hunt group supervisors can be off-node devices.*) Hunt groups can have one or more phones assigned as a hunt group supervisor. An extension list can be included in the list of supervisors. If desired, a phone can be assigned as the supervisor for more than one hunt group.

NOTE	A SIP device cannot be configured and used as a supervisor phone for ACD hunt groups.

If not using Agent IDs, ACD hunt group supervisors with display phones receive visual camp-on displays if they are programmed as members of the hunt group and they have the ACD Agent Logout feature enabled. If a Hunt Group is using ACD Agent IDs, the supervisor must be logged on to the group to receive camp-on indications. If not using ACD Agent IDs, you can add the supervisor as a hunt group member, set the Hunt Group flag for the supervisor's station to "Remove," and the supervisor will still receive hunt group camp-on indications.

To add supervisors to the hunt group:

1. Select System – Devices and Feature Codes – **Hunt Groups**.

2. Select the hunt group extension number.

3. Double-click **Supervisors** to see the current list of supervisors, if any.

4. Right-click anywhere in the right side of the window. An option box appears.

5. Select **Add To Supervisors List**. A window appears prompting for the device type to include.

6. Select the phone or extension types (you can use the SHIFT or CTRL key to select more than one item), and then click **Next**. The items with details appear. To view items in a list only, click **List**.

7. Select the appropriate items, then select **Add Items**. When you have added all the supervisors, click **Finish**. The selections appear in the list. To view programming options, double-click the extension number.

To delete supervisors:

1. Select System – Devices and Feature Codes – **Hunt Groups**.

2. Select the hunt group extension number.

3. Double-click **Supervisors** to see the current list of supervisors.

4. Select the supervisor(s) you want to remove and then press **DELETE** on the keyboard, or right-click and select **Remove Selected Items**.

TIMERS

You can program the following hunt group timers:

- **No Answer Advance**: The amount of time a call will ring at a hunt group phone (unanswered) before advancing to the next phone on the list. It is started when the call is received at the hunt group. The range is 1–255 seconds; the default value is 18 seconds.

- **Announcement**: *(UCD Hunt Groups Only)* The amount of time a call will remain unanswered before it is picked up by the hunt group announcement destination. It is started

when the call is received at the hunt group. The range is 1–255 seconds; the default value is 18 seconds.

- **Overflow**: *(UCD Hunt Groups Only)* The amount of time a call will circulate through the hunt group (unanswered) before being picked up by the hunt group overflow destination. This timer is started when the Announcement timer expires (or, if there is no announcement phone, when the call is received by the hunt group) and it is restarted each time the call leaves the overflow destination. The range is 1–255 seconds; the default value is 72 seconds.

- **Recall**: The amount of time a call will circulate through the hunt group (unanswered) before being sent to the hunt group recall destination. The timer is started when the call is received by the hunt group. The range is 1–65,535 seconds; the default value is 180 seconds (3 minutes).

- **Wrap-Up**: *(ACD Hunt Groups Only)* Each time an agent ends an ACD hunt group call, the ACD Wrap-Up Duration timer starts. Until the timer expires, the agent will not receive another call through any ACD hunt group. However, the agent can receive other non-ACD hunt group calls, direct ring-in calls, and transfers. The range is 1– 65,535 seconds; the default value is 15 seconds.

- **Average Connect Time Per Call**: *(UCD Hunt Groups Only)* An application announcement or overflow phone message can be programmed to include the caller's queue position and/ or estimated wait time. The estimated wait time is based on the Average Connect Time Per Call multiplied by the number of calls ahead of the caller in the queue, divided by the number of available hunt group members (*average connect time per call* x *number of waiting calls ÷ available members*). The range is 1–10,000 seconds; the default value is 60 seconds.

To program a hunt group timer:

1. Select System – Devices and Feature Codes – **Hunt Groups**.

2. Select the hunt group extension number.

3. Double-click **Timers**.

4. Select the timer, and then type the new value in the **Value** column.

5. Press **ENTER** or select another field to save the change.

ACD AGENT NO ANSWER – DND MESSAGE ADDITIONAL TEXT

This option is disabled and displays a red "X" if the ACD Hunt Group option is set to No, or if the ACD Agent No Answer – DND Message Number option is set at 0.

This option enables phone users to enter additional DND text when using the DND message chosen for the "DND Message Number" flag (see the following section).

> **NOTE** To determine if trunk calls will be allowed to go to DND for the selected hunt group, program the "Return ACD Calls to Hunt Group" option (see **page 584**).

To change the ACD Agent No Answer – DND Additional Text option:

1. Select System – Devices and Feature Codes – **Hunt Groups**.

2. Select the hunt group extension number.

3. Make sure the **ACD Hunt Group** option is set to **Yes** and that the **ACD Agent No Answer – DND Message Number** has a value other than **0**.

4. Enter the text, up to 16 characters, that displays when phone users select the ACD Agent No Answer – DND Message Number, or leave the field blank to allow users to enter their own text.

5. Press **Enter** or click outside of the field to save your changes.

ACD AGENT NO ANSWER – DND MESSAGE NUMBER

This option is disabled and display a red "X" if the "ACD Hunt Group" option is set to No.

if the last agent in a Hunt Group does not answer an incoming call, the agent is automatically placed in DND, and the call returns to the Hunt Group where it is answered or camped on. This flag and the "ACD Agent No Answer - DND Additional Text" flag (see the previous section) allow you to determine which DND message is used (selected by number) and the bottom line of the DND message text. This flag allows OAI applications to function better with the Hunt Groups.

The DND Message Number flag is a link to Phone-Related Information\Messages (see **page 464**) where you can select the language that you want to use and see which DND message you want to use. Phone users can select 0 – 20, which corresponds with the DND messages. A value of zero does not allow an agent to go into DND. It is set to zero by default.

> **NOTE** To determine if trunk calls will be allowed to go to DND for the selected hunt group, program the "Return ACD Calls to Hunt Group" option (see **page 584**).

To select the ACD Agent No Answer – DND Message Number:

1. Select System – Devices and Feature Codes – **Hunt Groups**.

2. Select the hunt group extension number.

3. In the Value column, select the number to be used for the **ACD Agent No Answer – DND Message Number**. You are prompted to view the Messages folder.

4. Do one of the following:
   - Click **Yes** to see the Messages folder and select the appropriate language. Click **DND** to see a list of DND messages from which you can determine the number to select. Then click the **Back** button on the toolbar several times to return to the Hunt Group folder to enter the value.
   - Click **No** to ignore the prompt and use the value you selected.

5. Click the **Value** column for **ACD Agent No Answer - DND Additional Text** to enter any additional text you want to accompany the DND message you chose in step 2.

6. Press **Enter** or click outside of the field to save your changes.

ACD HUNT GROUP

(This option is not recommended for Analog Voice Mail hunt groups.) This option allows the hunt group to use the ACD features. The ACD Hunt Groups software license, part no. 840.0230, is required to enable this option.

Automatic Call Distribution (ACD) hunt groups can use all of the standard and UCD hunt group features, if enabled, in addition to the following features:

- ACD hunt groups can be programmed to distribute calls to equalize call time or call count among the available members.

- ACD can provide call information records that can be processed by an external device connected to a system serial port.

- ACD hunt groups can use of Agent ID numbers in place of phone extensions in the hunt group list. See the Agent ID information on **"Programming ACD Hunt Groups"** on **page 569**.

To enable ACD hunt group features:

1. Select System – Devices and Feature Codes – **Hunt Groups**.

2. Select the hunt group extension number.

3. Select **ACD Hunt Group**.

4. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

5. Press **ENTER** or select another field to save the change.

ANALOG VOICE MAIL HUNT GROUP

If the Analog Voice Mail Hunt Groups software license, part no. 840.0229, is installed, the hunt group can be composed of phone ports which are connected to an analog voice mail device. Enabling this option has the effect of passing along phone identification when a call reaches the hunt group as a result of forwards and transfers. The purpose of this flag is to provide compatibility between the system and analog voice mail units. The optional external voice processing system is a digital system and does not require this flag.

NOTE	When an analog voice mail unit is connected to the system, the voice mail unit should use the Silent Message feature code (367) instead of the Message feature code (365). Mitel recommends *not* using this option with ACD hunt groups

To enable the Analog Voice Mail Hunt Group option:

1. Select System – Devices and Feature Codes – **Hunt Groups**.

2. Select the hunt group extension number.

3. Select **Analog Voice Mail Hunt Group.**

4. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

5. Press **ENTER** or select another field to save the change.

ANNOUNCEMENT AND OVERFLOW STATIONS

(*Announcement and overflow station phones can be assigned to off-node phones or Voice Processor applications, if they are programmed as off-node devices. See* **"Creating Off-Node Devices" on page 367**.

Hunt groups can have an announcement destination and/or an overflow destination. If a call to the hunt group is not answered before the Announcement timer expires, the call is picked up by the announcement destination. If the call remains unanswered when the hunt group Overflow timer expires, the call is picked up by the overflow destination. Announcement and overflow destinations should be playback device phones (these can be Auto Attendant, Voice Mail, or Call Routing Announcement applications). Do not include these applications in the hunt group distribution list.

To program an Announcement or Overflow phone:

Do one of the following:

Method A

    a. Select System – Devices and Feature Codes – **Hunt Groups**.

    b. Select the hunt group extension number.

    c. Select **Announcement**.

    d. Select the current value, and then enter the new value in the text box.

    e. Press **ENTER**. A screen appears displaying what is associated with the number entered.

    f. Click **OK**. The new number appears in the field.

Method B

    a. Select System Devices and Feature Codes – **Hunt Groups**.

    b. Select the hunt group extension number.

    c. Right-click the existing value for the Announcement or Overflow destination, and then click **Change Announcement**. The Change Announcement dialog box appears.

    d. Select the device types (you can use the SHIFT or CTRL key to select more than one item), and then click **Next**. The items with details appear. To view items in a list only, click **List**.

    e. Select the device you want as the Announcement or Overflow destination, and then click **Finish**. The selection appears in the applicable field.

AUDIO FOR CALLS CAMPED ONTO THIS DEVICE

The Audio for Calls Camped onto this Device field defines the audio that a caller hears when camped-on to this hunt group.

To program Audio for Calls Camped onto this Device:

1. Select System – Devices and Feature Codes – **Hunt Groups**.

2. Select the hunt group extension number.

3. Select **Audio for Calls Camped onto this Device**.

4. In the **Value** column, select one of the following options from the list:

    • **Silence**: Callers hear no Music-On-Hold.

- **Tick Tone**: Callers hear tick tone.

- **Ringback**: Callers hear ringback.

- **MOH Port**: Callers hear an external music source. This is the default value.

- **File-Based MOH**: Callers hear the MOH file selected in DB Programming. For more information, see "File-Based Music-On-Hold (MOH)" on **page 741**.

5. Press **ENTER** or click outside the field to save the change.

AUDIO FOR CALLS RINGING THIS DEVICE

The Audio for Calls Ringing this Device field defines the audio that a caller hears when ringing this hunt group. By default, the system determines the music source based on the trunk group in which the call resides. However, phones (as well as Hunt Groups and Voice Processor Applications) can be programmed to determine the music source a caller hears based on the device for which the caller is ringing.

NOTE	The Audio for Calls Ringing this Device option only works when the call goes through a trunk group and also when used in conjunction with the Use Next Device's Audio Source field. IC calls do not apply to the use of this field when this field is set to a music source. For a hunt group in which the primary purpose is to support IC callers (for example, an internal help desk), you should set all of the "Audio for Calls..." fields to something other than a music source, such as Ringback.

To program Audio for Calls Ringing this Device:

1. Select System – Devices and Feature Codes – **Hunt Groups**.

2. Select the hunt group extension number.

3. Select **Audio for Calls Camped onto this Device**.

4. In the **Value** column, select one of the following options from the list:

- **Silence**: Callers hear no Music-On-Hold.

- **Tick Tone**: Callers hear tick tone.

- **Ringback**: Callers hear ringback. This is the default value.

- **MOH Port**: Callers hear an external music source.

- **File-Based MOH**: Callers hear the MOH file selected in DB Programming. For more information, see "File-Based Music-On-Hold (MOH)" on **page 741**

5. Press **ENTER** or click outside the field to save the change.

AUDIO FOR CAMPED-ON ANNOUNCEMENT CALLS

This flag determines the functionality for camped-on hunt group calls that are listening to an Announcement Destination message when they transition from camped-on to ringing. If this flag is set for ringback, the user switches from hearing the Announcement Destination to hearing ringback when the call moves from camped-on to ringing. This is the default functionality.

If this flag is set for Announcement Destination, the user continues to hear the Announcement Destination when the call moves from camped-on to ringing. When the Announcement Destination hangs up (for example, the entire message is played), the user starts to hear ringback. If, however, an agent answers the call before the Announcement Destination hangs

up, the call is connected, and the rest of the message is not played. The user may hear Music-On-Hold instead of ringback depending on how the system is programmed.

To program Audio for Camped-On Announcement Calls:

1. Select System – Devices and Feature Codes – **Hunt Groups**.

2. Select the hunt group extension number.

3. Select **Audio for Camped-On Announcement Calls**.

4. In the **Value** column, select one of the following options from the list:
   - **Ringback**: Callers hear ringback. This is the default value.
   - **Announcement Destination**: Callers continue to hear the Announcement destination when the call moves from camped-on to ringing.

5. Press **ENTER** or click outside the field to save the change.

CAMP-ONS ALLOWED

If this flag is enabled, callers are allowed to camp on to the hunt group when all members are busy. If the flag is disabled, the callers hear busy tones when no members are available. In the default state, camp-ons are enabled.

To prevent callers from camping on to the hunt group:

1. Select System – Devices and Feature Codes – **Hunt Groups**.

2. Select the hunt group extension number.

3. Select **Camp-Ons Allowed**.

4. In the **Value** column, clear the check box. The field changes to **No**. To enable the option, select the check box.

5. Press **ENTER** or select another field to save the change.

GROUP CALL PICK-UP

The Group Call Pick-up feature allows users to reverse transfer a call that is ringing in to a hunt group or one of its phones using the hunt group extension number. For details, see "Group Call Pick-Up" on **page 568** and "Transfer – Reverse Transfer" on **page 339**.

- *If enabled,* users can enter the Reverse Transfer feature code (4), and then dial the hunt group extension number to pick up a call that is ringing in to any phone extension within that hunt group or to the hunt group extension number.

- If *disabled*, reverse transfers using the hunt group extension number will reverse transfer only calls ringing to the hunt group extension number, not calls ringing at the individual phones within the hunt group.

To enable Group Call Pick-Up:

1. Select System – Devices and Feature Codes – **Hunt Groups**.

2. Select the hunt group extension number.

3. Select **Group Call Pick-Up**.

4. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

5. Press **ENTER** or click out of the field to save the change.

PRIORITY LEVEL

Some phones may be members of more than one ACD or UCD hunt group. For this reason, hunt groups are assigned a "priority level." The priority level determines which hunt group calls should be received first when calls ring in and/or camp on to several hunt groups at once. If a phone is a member of multiple hunt groups that have the same priority level, calls received by those hunt groups will be queued in the order they were received by the system.

To set the hunt group priority level:

1. Select System – Devices and Feature Codes – **Hunt Groups**.

2. Select the hunt group extension number.

3. Select **Priority Level**.

4. In the **Value** column, select or type the priority level for this hunt group. Higher numbers have higher priority. The range is 0–75; the default is 0.

5. Press **ENTER** or click out of the field to save the change.

RECALL DESTINATION PHONE

The recall destination can be assigned to an off-node device, group, or application, if it is programmed as an off-node device. *S*ee "Creating Off-Node Devices" on **page 367**. Any hunt group can have a recall destination, regardless of whether UCD is enabled. If a call remains unanswered when the Recall timer expires, it will be sent to the recall destination. Do not include this device in the hunt group distribution list.

To program the Recall Destination:

Use one of the following methods:

Method A

a. Select System – Devices and Feature Codes – **Hunt Groups**.

b. Select the hunt group extension number.

c. Select **Recall**.

d. In the **Value** column, type the new destination number.

e. Press **ENTER** or click out of the field to save the change. A screen appears showing what is associated with the number entered.

f. Click **OK**. The new number appears in the field.

Method B

a. Select System – Devices and Feature Codes – **Hunt Groups**.

b. Select the hunt group extension number.

c. Select **Recall**.

d. Right-click the existing value for the Recall phone, and then click **Change Recall**. The Change Recall dialog box appears.

e. Select the device types (you can use the SHIFT or CTRL key to select more than one item), and then click **Next**. The items with details appear. To view items in a list only, click **List**.

f. Select the device you want as the Recall destination, and then click **Finish**. The selection appears in the Recall field.

RESTART ACD IDLE TIME UPON LOGIN

This option determines where an agent is placed in a longest idle queue when the agent logs back in to a hunt group.

- When *enabled*, the agent's idle time is reset to zero whenever the agent logs in (for example, that agent will be least likely to receive the next distributed call).

- When *disabled*, the agent's idle time includes the time the agent was logged out of the hunt group (for example, that agent will be most likely to receive the next distributed call). By default, this flag is disabled.

To enable the Restart ACD Idle Time Upon Login option:

1. Select System – Devices and Feature Codes – **Hunt Groups**.

2. Select the hunt group extension number.

3. Select **Restart ACD Idle Time Upon Login**.

4. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

5. Press **ENTER** or click out of the field to save the change.

RETURN ACD CALLS TO HUNT GROUP

When enabled, this option re-queues a call to the front of the Camp On queue for the hunt group from which the call came. It allows a calling party to immediately return to the front of the hunt group queue if the ringing agent enters Do-Not-Disturb (DND) mode. If the option is not enabled, the calling party will continue to ring the agent until the No Answer Advance timer expires or the agent answers the call.

To enable the Return ACD Calls to Hunt Group option:

1. Select System – Devices and Feature Codes – **Hunt Groups**.

2. Select the hunt group extension number.

3. Select **Return ACD Calls to Hunt Group**.

4. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

5. Press **ENTER** or click out of the field to save the change.

SEARCH TYPE

Available Search Types depend on the type of hunt group being programmed:

- **For Standard and UCD Hunt Groups**: Determine whether the calls are sent to the phones in linear or distributed order. Linear order means that the call is sent to the first phone or extension list on the list and moves down the list until it reaches an available phone. With distributed order, the call is sent to the phone that appears on the list after the last phone or extension list to receive a call (even if the call was not answered).

- **For ACD Hunt Groups**: ACD Hunt Group calls can circulate in linear or distributed order (as described above) or using one of the ACD distribution methods: Longest Idle or Balance Call Count order. Longest idle means that an incoming call is sent to the phone that has not been involved in a call to this hunt group for the longest period of time. Balance Call Count means that, to balance the call load, each incoming call is sent to the phone that has received the fewest calls through this hunt group. (It does not count calls that were received through other hunt groups, direct ring-in, or transfer.)

To set the search type:

1. Select System – Devices and Feature Codes – **Hunt Groups**.

2. Select the hunt group extension number.

3. Select **Search Type**.

4. In the **Value** column, select the search type from the list.

5. Press **ENTER** or click out of the field to save the change.

SEND CAMP ON NOTIFICATIONS TO MEMBERS IN DND

This option indicates whether or not camp-on burst tones are sent to hunt group members that are in DND or that are logged out. When sent to display phones, the display shows N CALLS WAITING FOR <HUNT GROUP>, where *N* is the number of calls that are currently camped-on to the hunt group. This allows hunt group members to see the hunt group queue in real-time.

When this flag is enabled:

- If the hunt group is a UCD hunt group, members receive one of the following notifications:
    - Logged into hunt group and in DND: Tone -YES
    - Logged into hunt group and not in DND: Tone - YES
    - Logged out of hunt group and in DND: Tone - YES
    - Logged out of hunt group and not in DND: Tone - YES
- If the hunt group in an ACD hunt group but does *not* use Agent IDs, members receive one of the following notifications:
    - Logged into hunt group and in DND: Tone -YES
    - Logged into hunt group and not in DND: Tone - YES
    - Logged out of hunt group and in DND: Tone - YES
    - Logged out of hunt group and not in DND: Tone - YES
- If the hunt group is an ACD hunt group and uses ACD Agent IDs:

- The user receives Camp On notifications if the Agent ID is **logged in** to the Hunt Group *and* in DND.

- The user does *not* receive Camp On notifications if the Agent ID is **logged out** of the hunt group.

When this flag is disabled:

- If the hunt group is a UCD hunt group, members receive one of the following notifications:
    - Logged out of hunt group and not in DND: Tone - YES
    - Logged out of hunt group and in DND: Tone - NO
    - Logged into hunt group and not in DND: Tone - YES
    - Logged into hunt group and in DND: Tone - NO

- If the hunt group in an ACD hunt group but does *not* use Agent IDs, members receive one of the following notifications:
    - Logged out of hunt group and not in DND: Tone - YES
    - Logged out of hunt group and in DND: Tone - NO
    - Logged into hunt group and not in DND: Tone - YES
    - Logged into hunt group and in DND: Tone - NO

- If the hunt group is an ACD hunt group and uses ACD Agent IDs:
    - The user does *not* receive Camp On notifications if the Agent ID is **logged in** to the Hunt Group *and* in DND.
    - The user does *not* receive Camp On notifications if the Agent ID is **logged out** of the hunt group.

To enable the Send Camp On Notifications to Members in DND option:

1. Select System – Devices and Feature Codes – **Hunt Groups**.

2. Select the hunt group extension number.

3. Select **Send Camp On Notifications to Members in DND**.

4. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

5. Press **ENTER** or click out of the field to save the change.

USE ACD AGENT IDS

If the ACD Hunt Group Option is enabled, you can choose to route calls according to ACD Agent ID numbers instead of phone extensions.

To enable the Use ACD Agent IDs option:

1. Select System – Devices and Feature Codes – **Hunt Groups**.

2. Select the hunt group extension number.

3. Select **Use ACD Agent IDs**.

4. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

5. Press **ENTER** or click out of the field to save the change.

NODE-SPANNING HUNT GROUPS

There is a software license called Remote Automatic Call Distribution Hunt Groups. It allows ACD hunt groups to span nodes. Node-spanning ACD hunt groups can have either members or ACD Agent IDs. Agent IDs are considered global throughout all nodes in which the Agent exists. This means that if you create Agent ID 100 on one node, you must also create Agent ID 100 on all other nodes that have a phone that may want to login to the hunt group using that ID.

PROGRAMMING REMOTE (OFF-NODE) HUNT GROUPS

When you double-click a remote node, a list of the existing off-node hunt groups for that node, if any, appears. You can create or delete off-node hunt groups or enter descriptions and user names.

To create an off-node hunt group:

1. Right-click anywhere in the right pane, and then select **Create Off-Node Hunt Group**. The Create Hunt Group dialog box appears.

2. Enter the extension number in the box or scroll to the desired number.

3. Click **OK** to continue. The new hunt group appears in the list without a description or username.

4. Program the **Description** and **Username fields**. All hunt groups should have a description and a username. The description appears in all hunt group lists in the database and can be up to 20 characters long. The username appears on phone displays and can have up to 10 characters.

    To program the Description and Username fields:

    a. Select the box.

    b. Type the entry.

    c. Press **ENTER** or click out of the field to save your changes.

NETWORK GROUPS

This section contains the following information:

- **Feature Description** below

- Programming Network Groups on **page 591**

FEATURE DESCRIPTION

Network Groups define the IP devices that can communicate through Peer-to-Peer (P2P) media. When you assign an IP device (including SIP phones and MGCP gateways and phones) to a circuit, the device is automatically added to the default Network Group (PP029). This Network Group is not programmed for P2P media and it cannot be configured. To use P2P media, you must assign the IP devices to a Network Group that is programmed to support the feature. Before you assign IP devices to a Network Group, however, you must make sure the hardware is properly upgraded. You cannot delete the default Network Group. For more information about Network Groups, see "P2P Network Groups" on **page 590**.

For a more current list of compatible SIP gateways, refer to KB article 09-4940-00056 on the Mitel Knowledge Base (KB) Center (http://domino1.mitel.com/prodsupp/prodsupkb.nsf/WebSearchForm?OpenForm). Note that you must first log on to Mitel OnLine (http://portal.mitel.com/wps/myportal/MOLHome) in order to access the KB.

The P2P media feature allows certain Internet Protocol (IP) and multi-protocol (SIP or ITP) devices to transmit and receive audio directly with each other. With this feature, the audio is not transmitted or received through the system chassis. This reduces delay and removes the audio stream from the Time Division Multiplex (TDM) highway.

> **NOTICE**
>
> Passing real-time streaming data, such as audio, through encrypted virtual private networks (VPN) may significantly impact network performance, router and firewall functionality, and audio quality.

Devices that support P2P media include:

- A SIP gateway and associated phones

- An MGCP gateway and associated phones

- IP phones

- SIP phones

When using P2P media:

- The devices must be members of the same Network Group.

- A Network Address Translation (NAT) device or firewall cannot exist between the communicating phones. For detailed information about the configuration of NAT devices in a network environment, refer to "Appendix B: Network IP Topology," in the *MiVoice Office 250 Installation Manual* .

Because the devices are not using unit chassis resources, devices connected in a P2P call cannot use the following features:

- Agent Help

- Record-A-Call

- Station Monitor

HARDWARE UPGRADES

To have IP phones use P2P to communicate through SIP/MGCP gateways, place the IP phones and SIP/MGCP trunks (gateway and phones) in the same Network Group. The P2P option must be set on that Network Group. P2P media requires upgrading IP phones and IP SLAs to the latest firmware version.

NOTICE

Run the Network Qualifier (for at least 24 hours) to determine if the network meets the minimum requirements identified for P2P media. These requirements are the same as those identified for IP networking and IP private networking. For more information about the Network Qualifier, refer to the *Network Qualifier Reference Manual*.

To upgrade the hardware:

Use the Upload Utility or TFTP to upload the latest firmware to all IP devices that use P2P media. For information on using TFTP to upgrade IP devices, refer to the *MiVoice Office 250 Installation Manual* .

REQUIREMENTS AND CONSTRAINTS

The following are requirements and constraints when using P2P media:

- The phones must be members of the same Network Group, and the Network Group flag must be set to True. The default Network Group cannot operate in P2P mode, but any other Network Group can be programmed for P2P operation.

- IP devices must be upgraded to the latest version of firmware. SIP phones do not require firmware upgrades.

- A Network Address Translation (NAT) device or firewall cannot be placed between the communicating phones.

Because the devices are not using chassis resources, phones connected in a P2P call cannot use the following features:

- Agent Help

- Record-A-Call

- Station Monitor

P2P media requires that the user upgrade IP phones and IP SLAs to the latest version of firmware.

NOTE | Mitel recommends running the Network Qualifier for at least 24 hours to determine if the network meets the minimum requirements identified for P2P media.

P2P NETWORK GROUPS

Network Groups define the IP devices, SIP phones, MGCP phones, and SIP gateway trunks that can connect to each other using P2P media. Only devices within the same Network Group can talk to each other using P2P media. Even devices that are on separate nodes must be in the same Network Group. For example, for devices on Node 2 to communicate in P2P mode with devices on Node 3, both nodes must belong to the same Network Group, with the same extension programmed. The associated devices on each node must then be members of that group. If two devices are in separate Network Groups, P2P is not used, and the call is routed through the chassis.

SIP PEER-TO-PEER

A SIP Peer can now be put into a Network Group that supports peer-to-peer media. The Peer-to-Peer (P2P) functionality enables the new SIP Peer devices to send and receive audio packets directly to the other IP device (without backplane connections). This implementation is the same as the existing 5000 IP devices.

- Each SIP Peer is a member of a Network Group.

- The Network Group may allow P2P media.

- If P2P media is enabled, any devices in the group will try to negotiate P2P media streams during establishment of an audio connection. If the devices are unable to negotiate a P2P stream, they will fall back to using backplane connections.

Most call scenarios will attempt to negotiate a P2P stream except the following:

- An Ad-Hoc Conference call that breaks down to two-parties

- DEE calls that route to the external destination and Mid-Call Features are enabled

- DEE calls that route to the external destination (e.g., Mobile phone) during Human Answer Supervision

- The two devices are in different Network Groups.

The following also affect SIP P2P functionality:

- A call that is using P2P media cannot use Record-A-Call or be monitored using by Station Monitor.

- SIP P2P functionality can work across nodes. However, the extensions of the Network Groups must match and the P2P flag needs to be enabled on both nodes.

- SIP P2P functionality is backwards compatible with older MiVoice Office 250 versions. However, Mitel recommends upgrading all nodes to the latest 5000 versions when deploying a multi-node system.

- SIP P2P functionality can work with an MCD using Private Networking via SIP Peer Trunk Groups.

When placing SIP devices in a Network Group and enabling P2P, it should be noted that the 5000 assumes that all devices in the same Network Group can use the same set of codecs (e.g., G.711 and G.729). If a Network Group contains two devices that use different codecs, for example, one can only use G.711 and the other G.729, then this call can results in no audio. In this case, P2P media should not be configured.

To add a SIP Phone Groups to a Network Group:

1. Go to **System\Devices and Feature Codes\Network Group\<extension>** folder.

2. Click **SIP Phone Groups**.

3. Right-click anywhere in the right-side pane.

4. Select **Move to SIP Phone Groups List**.
    The *Move to SIP Phone Groups List* dialog displays.

5. Select **SIP Phone Groups** and click **Next**.

6. Select the devices to add to the list and click **Move Items**.

7. Click **Finish** when all of the desired devices have been moved.

To add a SIP Trunk Groups to a Network Group:

1. Go to **System\Devices and Feature Codes\Network Group\<extension>** folder.

2. Click **SIP Trunk Groups**.

3. Right-click anywhere in the right-side pane.

4. Select **Move to SIP Trunk Groups List**.
    The *Move to SIP Trunk Groups List* dialog displays.

5. Select **SIP Trunk Groups** and click **Next**.

6. Select the devices to add to the list and click **Move Items**.

7. Click **Finish** when all of the desired devices have been moved.

To add a SIP Voice Mail Groups to a Network Group:

1. Go to **System\Devices and Feature Codes\Network Group\<extension>** folder.

2. Click **SIP Voice Mail Groups**.

3. Right-click anywhere in the right-side pane.

4. Select **Move to SIP Voice Mail Groups List**.
    The *Move to SIP Voice Mail Groups List* dialog displays.

5. Select **SIP Voice Mail Groups** and click **Next**.

6. Select the devices to add to the list and click **Move Items**.

7. Click **Finish** when all of the desired devices have been moved.

PROGRAMMING NETWORK GROUPS

This section contains the following information:

*   **Creating Network Groups** below

*   **Creating Network Group Phones and Trunks** below

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CREATING NETWORK GROUPS

When you double-click Network Groups, the default Network Group (PP029) and any other programmed groups are displayed.

NOTE	IP softphones that are used as mobile devices cannot be part of a Network Group configured for P2P media. If you attempt to add an IP softphone to the Network Group list or change the **Use Peer-to-Peer Media** setting of a Network Group to **Yes**, a warning message appears.

To create a Network Group:

1. Select System – Devices and Feature Codes – Network Groups.

2. Right-click, and then select **Create Network Group**.

3. Select a P7XXX extension and the number of groups to create.

4. Click **OK**. Each Network Group on a node must have a unique extension.

5. Program the description and user name. The description that appears can be up to 20 characters long. The username, that will appear on display phones, can have up to eight characters. Do not use slash (/), backslash (\), vertical slash (|), or tilde (~) characters in usernames. Do not use Control characters in descriptions or usernames.

6. In the **Use Peer-to-Peer Media** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box. If this option is disabled, the devices in the group use the system vocoders for all calls.

CREATING NETWORK GROUP PHONES AND TRUNKS

After you have created the Network Group, you must assign phones and trunks to the group. The phones and/or trunks in this group will then use P2P media or the TDM highway based on the **Use Peer-to-Peer Media** option (see step 6 on **page 592**).

Network Group IP Phones

IP phones include IP phones (except the IP SoftPhone), IP SLAs, and multi-protocol (SIP or IP) phones. To view a list of IP phones that are currently assigned to the Network Group, double-click **Phones**.

> ### NOTICE
>
> **8602 softphones are not supported for P2P media and network groups.** 8602 IP softphone applications may be used as mobile devices, allowing them to move between the LAN containing a MiVoice Office 250 and public network/Internet. However, any 8602 used for this purpose must *not* be included in a Network Group using P2P media.

To add phones to the list:

1. Right-click and select **Move To Phones List**.

2. Select IP Phone or IP Single-Line Adapter, and then click **Next**.

3. Select the devices to add to the list, then click **Move To List**.

4. Click **Finish** when all devices have been moved.

Network Group Trunks

IP trunks include SIP/MGCP gateways and phones. To view a list of IP trunks that are currently assigned to the Network Group, double-click **Trunks**.

To add trunks to the list:

1. Right-click and select **Move To Trunks List**.

2. Select **MGCP Phone** or **MGCP Gateway and Phone** or **SIP Trunk**, and then click **Next**.

3. Select the devices to add to the list, and then click **Move To List**.

4. Click **Finish** when all devices have been moved.

You can also program Network Groups for P2P media across nodes, as described below.

To program Network Groups for P2P media across nodes:

1. Create a Network Group on each node and assign each group the same extension number. For example, if the devices that will use P2P media reside on nodes 2 and 3, program a Network Group with the same extension number (for example, P7000) on both nodes.

2. Assign the appropriate phones and trunks to the Network Group on each node. For example, to use P2P media between device 2000 on node 2 and device 3000 on node 3, Network Group P7000 on node 2 must contain device 2000, and Network Group P7000 on node 3 must contain device 3000.

3. Enable the **Use Peer-to-Peer Media** flag for the Network Group on each node.

NODE IP CONNECTION GROUPS FOR REMOTE NODES

The system automatically creates a node IP connection group for each remote node added, as shown in the figure below.

The node IP connection group, which corresponds to an IP network connection between the remote node and the local node, can then be programmed for each off-node IP connection. Like other devices, DB Programming displays off-node IP connections within a folder corresponding to the node on which the off-node IP connection resides. When you access the IP Connections folder, the right side of the screen displays the extension, description, and username of the IP connection group.

All IP connection groups should have a description and a username. The description that appears in all IP connection group lists in the database can be up to 20 characters long. The username, that will appear on display phones, can have up to 10 characters. To program the names, select the desired text box and type the entry. Do not use slash (/), backslash (\), vertical slash (|), or tilde (~) characters in usernames. Do not use Control characters in descriptions or usernames.

You can program the following IP Call Configuration options:

- Node IP Connection Group IP Call Configurations on **page 595**

- Day/Night Emergency Outgoing Access on **page 596**

- Day or Night Outgoing Access on **page 510**

- Remote Node on **page 597**

- Camp-Ons Allowed on **page 598**

NODE IP CONNECTION GROUP IP CALL CONFIGURATIONS

Changes to IP Call Configuration settings do not take effect until the next call is placed or received. Changes do not affect existing calls. For descriptions about IP Call configuration options, see "Programming Call Configuration Options" on **page 632**.

To program a Node IP Connection Group IP Call Configuration:

1. Select System – Devices and Feature Codes – Node IP Connection Groups – <***Node Connection Group number***>.

2. Double-click **IP Call Configuration**. IP Call Configuration options are shown in the right pane.

LOCAL MUSIC SOURCE

Available only when you set the Music-On-Hold vocoder to Local Music Source. When set to Local Music Source, the system does not transmit and receive Music-On-Hold between the local node and the remote node. Rather, the local music source field defines the music source the caller on the local node hears when he would otherwise be listening to Music-On-Hold.

To program the Local Music Source:

1. Select System – Devices and Feature Codes – Node IP Connection Groups – <***Node Connection Group number***>.

2. Select the call configuration.

3. Select **Local Music Source**.

4. In the **Value** column, select one of the following options from the list:

   - **Silence**: Callers hear no Music-On-Hold.

   - **Tick Tone**: Callers hear tick tone.

   - **Ringback**: Callers hear ringback.

   - **MiVoice Office 250**: Callers hear an external music source (this is the default value).

   - **File-Based MOH**: Callers hear the MOH file selected in DB Programming. For more information, see "File-Based Music-On-Hold (MOH)" on **page 741**.

5. Press **ENTER** or click outside the field to save the change.

MUSIC-ON-HOLD ENCODING SETTING

Determines whether Music-On-Hold is transmitted across a network connection. You can program this field to either *Use Speech Encoding Setting* or *Use Local Music Source.* If set to *Use Speech Encoding Setting,* the system uses the vocoder specified by the speech encoding setting when transmitting Music-On-Hold across the network connection. If set to *Use Local Music Source*, the system connects the caller to the music source given by the local music field rather than transmitting Music-On-Hold across the network.

To select the Music-On-Hold Encoding Setting:

1. Select System – Devices and Feature Codes – Node IP Connection Groups – <***Node Connection Group number***>.

2. Select the call configuration.

3. Select **Music-On-Hold Encoding Setting**.

4. In the **Value** column, select the option from the list. The default is Use Local Music Source.

5. Press **ENTER** or click out of the field to save the change.

DAY/NIGHT EMERGENCY OUTGOING ACCESS

There are separate lists for phones with emergency outgoing access in day and night modes. By default, the automatic phone list (Auto: All Phones) is assigned to Day/Night Emergency Outgoing Access.

To add phones having emergency outgoing access for a node trunk group:

1. Select System – Devices and Feature Codes – Node IP Connection Groups – <***Node Connection Group number***>.

2. Select **Emergency Outgoing Access**. The choices are Day Mode or Night Mode.

3. Right-click anywhere in the right side of the window. An option box appears.

4. Select **Add To List**. A window appears prompting for the device type to include.

5. Select the device types (you can use the SHIFT or CTRL key to select more than one item), and then click **Next**. The items with details appear. To view items in a list only, click **List**.

6. Select the appropriate items, and then select **Add Items**.

7. When you have added all the desired devices, click **Finish**. The selections appear in the list. To view programming options, double-click the extension number.

DAY/NIGHT OUTGOING ACCESS

There are separate lists for phones with outgoing access in day and night modes. By default, the automatic phone list (*Auto: All Phones*) is assigned to Day/Night Outgoing Access.

To add phones have outgoing access for a node trunk group:

1. Select System – Devices and Feature Codes – Node IP Connection Groups – <***Node Connection Group number***>.

2. Select **Outgoing Access**, and then double-click **Day Mode** or **Night Mode**.

3. Right-click anywhere in the right pane, and then click **Add to Day** (or **Night**) **List**.

4. Select **Add To List**. The Add to Day (or Night) dialog box appears.

5. Select the device types (you can use the SHIFT or CTRL key to select more than one item), and then click **Next**. The items with details appear. To view items in a list only, click **List**.

6. Select the appropriate items, and then select **Add Items**.

7. When you have added all the desired devices, click **Finish**. The selections appear in the list. To view programming options, double-click the device.

To remove phones that have outgoing access for the node trunk group:

1. Select System – Devices and Feature Codes – Node IP Connection Groups – <***Node Connection Group number***>.

2. Select **Outgoing Access**, and then select **Day Mode** or **Night Mode**.

3. Select the item, right-click, and then select **Remove Selected Items**.

To delete phone(s) having emergency outgoing access for a Node IP Connection group:

1. Select System – Devices and Feature Codes – Node IP Connection Groups – <***Node Connection Group number***>.

2. Select **Emergency Outgoing Access**.

3. Select **Day Mode** or **Night Mode**.

4. Select the phone(s) you want to remove, right-click, and then select **Remove Selected Items**.

NOTICE

Responsibility for Regulatory Compliance.

It is the responsibility of the organization and person(s) performing the installation and maintenance of Mitel Advanced Communications Platforms to know and comply with all regulations required for ensuring Emergency Outgoing Access at the location of both the main system and any remote communication phones. Remote IP and SIP phones may require gateway access to nearby emergency responders.

Emergency Call phone numbers include:

- 911, the default for Mitel systems located in the U.S.
- 999, the default for Mitel systems located in the European market and used primarily in the United Kingdom U.K.
- If applicable, 112, an emergency number used widely in Europe outside of the U.K.
- 112, the default for Mitel systems located in Australia.
- Any emergency number, such as for a police or fire station, that is appropriate for the location of the main system and/or remote phones.

REMOTE NODE

The Remote Node field is a link to the node to which the connection group corresponds.

CAMP-ONS ALLOWED

The Camp On feature can be enabled or disabled for each node IP connection group. If disabled, users placing outgoing calls will hear busy signals when all connections in the group are in use or unavailable. If enabled, users will be able to camp on and wait for an available connection.

To enable/disable the Camp On Allowed flag:

1. Select System – Devices and Feature Codes – Node IP Connection Groups – <***Node Connection Group number***>.

2. Select **Camp-Ons Allowed**.

3. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

4. Press **ENTER** or select another field to save the change.

Chapter 10

SYSTEM AND DEVICE IP SETTINGS

INTRODUCTION

This chapter describes MiVoice Office 250 Internet Protocol (IP) features and functionality. This includes 5000 system IP settings in the organizational network and phone IP settings and features.

System IP devices include the following:

- IP phones

- Session Initiated Protocol (SIP) phones

- MGCP gateways and phones

- SIP gateways and phones

For more information about system IP networks, refer to the following resources in the *MiVoice Office 250 Installation Manual* :

- Appendix A: Private Networking

- Appendix B: Network IP Topology

This chapter includes the following IP settings and devices information:

- IP Device Status on **page 601**

- System IP Settings on **page 603**

- Local Processor Module and Expansion Card IP Settings on **page 617**

- Remote Node IP Connections on **page 626**

- IP Call Configurations on **page 629**

- Sockets on **page 640**

- Phone and Device IP Settings on **page 641**

- Resource Reservation Tool on **page 647**

For IP device-related troubleshooting tips, see **page 657**.

IP DEVICE STATUS

When a phone comes online, it searches for a valid license. For more information about IP device licensing, refer to the System Description chapter in the *MiVoice Office 250 Installation Manual* .

Table 94:   IP Device Status Dialog Descriptions – License Category

ENTRY IN DIALOG	ENTRY DESCRIPTION
Unlicensed	The phone was unable to retrieve a valid license and is currently unlicensed.
Category A	The phone is using a Category A license.
Category B	The phone is using Category B license.
Category C	The phone is using a Category C license.
Category D	The phone is using a Category D license.
Category E	The phone is using a Category E license.
Category F	The phone is using a Category F license.

You can view the connection status of Mitel IIP devices on the system at any time from the IP Status page. The window displays the following information about each IP device.

• **Extension**: Indicates the extension number assigned to the IP device.

• **Type:** Displays the device type associated with the extension. Possible options include IP PHONE and IP SLA.

• **Network Group**: Displays the network group that the extension has been assigned. If a network group has not been assigned, the extension will reside in the default PP029.

• **DHCP**: Indicates whether the MiVoice Office 250 DB Programming is configured to DHCP for this device. It does not necessarily mean the device is actually using DHCP, the device could be set to a static IP.

• **Current IP Address**: Displays the IP address currently assigned to the device.

• **MAC Address**: Displays the MAC address assigned to the device.

• **Model Types**: Indicates the type of IP Device.

• **License Category**: Indicates the type of license associated with the phone.

The Status column in the IP Device Status option, as shown in the figure below, contains the following status indications for IP resource allocation:

• **Online**: Indicates that the device is currently online.

• **Offline**: Indicates that the device is currently offline.

• **No License**: Indicates that there is no license available for the device.

• **Resource In Use**: Indicates that the device has requested and has been allocated an IP resource.

•   **Waiting For Resource**: Indicates that the device has requested a resource, but all of the resources were in use. Therefore, the device is camped on to wait for a resource.

For more information about IP Device Status, refer to the *MiVoice Office 250 DB Programming Help.*

Status of
Resource Allocation

Get IP Device Status...

	Extension	Type	Network Group	DHCP	Current IP Address	MAC Address	Model Type	License Category	Status
	1000	86xx	PP029	Yes	172.17.158.148	00:10:36:01:FF:C8	8662	Category 'A' Phones	Resource In Us
	1004	86xx	PP029	Yes	192.168.200.201	00:10:36:00:FF:FF	None	None	Offline
	1006	86xx	PP029	Yes	192.168.200.201	00:10:36:11:FF:FF	None	None	Offline
	1010	86xx	PP029	Yes	192.168.200.201	00:10:36:00:DE:FF	None	None	Offline
	1200	86xx	PP029	Yes	192.168.200.201	00:10:36:05:5B:51	None	None	Offline
	1201	86xx	PP029	Yes	192.168.200.201	00:10:36:01:57:FF	None	None	Offline
	1300	86xx	PP029	Yes	192.168.200.201	00:10:36:03:AF:87	None	None	Offline
	1402	86xx	PP029	Yes	172.16.11.3	00:10:36:02:82:69	8662	Category 'A' Phones	Online
	1002	IP Softphone	PP029	Yes	192.168.200.201	11:11:11:11:11:12	None	None	Offline
	1301	IP Softphone	PP029	Yes	192.168.200.201	86:02:00:00:10:07	None	None	Offline
	1001	52xx/53xx	PP029	Yes	172.17.158.81	08:00:0F:2E:93:74	5224	Category 'D' Phones	Resource In Us
	1005	52xx/53xx	PP029	Yes	192.168.200.201	00:00:00:00:00:00	None	None	Offline
	1007	52xx/53xx	PP029	Yes	192.168.200.201	00:00:00:00:00:00	None	None	Offline
	1008	52xx/53xx	PP029	Yes	192.168.200.201	00:00:00:00:00:00	None	None	Offline
	1400	52xx/53xx	PP029	Yes	192.168.200.201	08:00:0F:26:55:CE	None	None	Offline
	1401	52xx/53xx	PP029	Yes	192.168.200.201	08:00:0F:43:A2:9E	None	None	Offline
	1500	52xx/53xx	PP029	Yes	192.168.200.201	08:00:0F:30:C5:3B	None	None	Offline
	94011	SIP TRUNK	PP029	N/A	192.168.200.201	N/A	None	None	Online
	94012	SIP TRUNK	PP029	N/A	192.168.200.201	N/A	None	None	Online
	94013	SIP TRUNK	PP029	N/A	192.168.200.201	N/A	None	None	Online

Status

Operation was a success.   IP Address: 172.16.10.178

[Get Status]   [Close]

SYSTEM IP SETTINGS

Changing IP database settings may drop all calls in progress.

This following sections describes IP settings, which includes the following:

- **General IP Settings Parameters**: General IP options that apply for the Base Server and the Processing Server. **For more information, see page 603.**

- **Base Server/Processing Server Connection Settings:** Contains settings required for communication between the Base Server and the Processing Server. For more information, see **page 611**.

> **NOTE** The Base Server\Processing Server Connection Settings option apply to systems equipped with a PS-1. If your system does not have a PS-1, this option and its settings appear with a red "X."

- **DHCP Server Settings: Options to configure the embedded DHCP server. For more information, see page 612.**

- **Web/SSH Settings:** Options to configure the SSH server or the Administrative Web Portal. **For more information, see page 613.**

- **Advanced IP Settings**: Options to configure the WINS and DNS servers. **For more information, see page 614.**

- **NTP Server Configuration Settings**: Options to configure the NTP Server. When the Enable Network Time Protocol (NTP) flag is set to No, these fields are displayed with a red "X." Every time this field is changed, the system attempts an NTP update. For more information, **see page 615.**

- **Remote Configuration Settings**: *This feature is reserved for controlled introduction.* Configures on-demand Remote Configuration options. For more information, see "" on **page 89**.

GENERAL IP SETTINGS

Table 95 on **page 605** shows general IP setting options that apply to the MiVoice Office 250.

To view or change General IP settings:

Select System – IP Settings. General IP settings are shown in the right pane, as shown below.
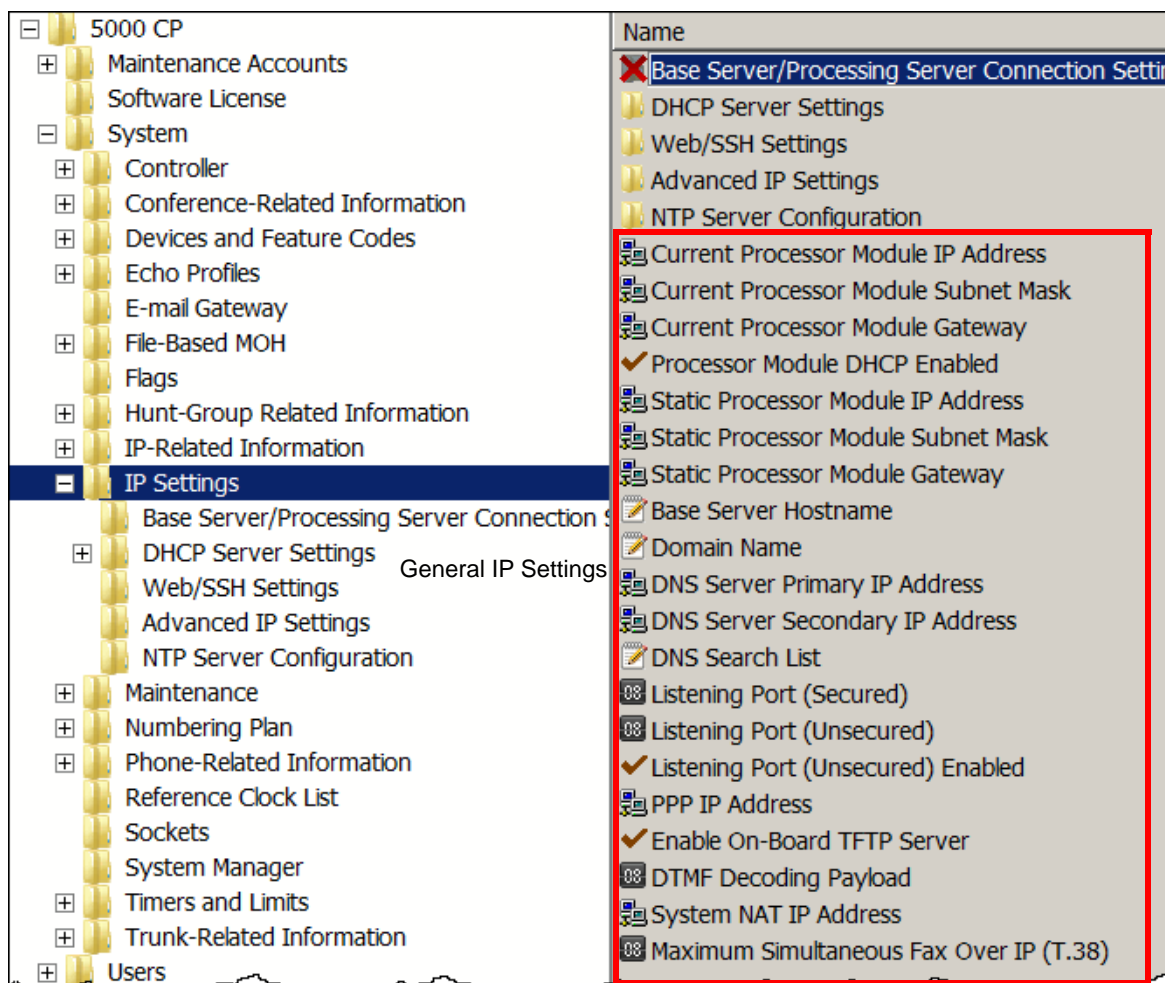
General IP Settings

Table 95 shows general IP setting fields.

| NOTES | The fields associated with the Processor Expansion Card appear only when there is a Processor Expansion Card in the system and the fields associated with the Processing Server appear only when there is a Processing Server in the system. |
| | The static fields associated with the Processing Server appear with a red "X" if DHCP is enabled for the Processing Server. The static fields associated with the Processor Expansion Card never appear with a red "X" (DHCP cannot be enabled for the Processor Expansion Card). |

Table 95: General IP Settings Fields

FIELD	DESCRIPTION	RANGE	DEFAULT
Current Processor Module IP Address	*Read-only.* Displays the current IP address of the Processor Module on the Base Server. This value may change if using a DHCP server to obtain an IP address.	0.0.0.0 – 255.255.255.255	192.168.200.201
Current Processor Module Subnet Mask	*Read-only.* Displays the current subnet mask associated with the Processor Module on the Base Server. This value may change if using a DHCP server to obtain an IP address.	0.0.0.0 – 255.255.255.255	255.255.255.0
Current Processor Module Gateway	*Read-only.* Displays the current gateway IP address associated with the Processor Module on the Base Server. This value may change if using a DHCP server to obtain an IP address.	0.0.0.0 – 255.255.255.255	192.168.200.1
Processor Module DHCP Enabled	Determines whether Dynamic Host Configuration Protocol (DHCP) is used for the Processor Module.	Yes/No	Yes
Static Processor Module IP Address	Specifies the static IP Address of the Processor Module on the Base Server. It appears with a red "X" if DHCP is enabled.	0.0.0.0 – 255.255.255.255	192.168.200.201
Static Processor Module Subnet Mask	Specifies the static subnet mask associated with the Processor Module. This setting should be provided by the IP network administrator. Changing this setting resets the IP resource application.	0.0.0.0 – 255.255.255.255	255.255.255.0
Static Processor Module Gateway	Specifies the static IP address of the gateway that is used to access the Processor Module IP port. Displays with a red "X" if DHCP is enabled.	0.0.0.0 – 255.255.255.255	192.168.200.1

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Table 95:   General IP Settings Fields (continued)

FIELD	DESCRIPTION	RANGE	DEFAULT
Base Server Hostname	Specifies the Base Server hostname provided by the IP network administrator. Allows you to access the system without having to enter the IP address. Also used to identify the system on the network. Because the Unified Voice Messaging (UVM) e-mail system requires DNS and a valid hostname, deleting the hostname causes UVM to fail. For more information about UVM, see "Mitel Voice Processing Systems" on **page 836**. **Note:** To receive and send e-mail messages using Voice Profile for Internet Mail (VPIM), the Base Server Hostname must be the same as the DNS hostname programmed in the Domain Name field under System – IP Settings.If the hostname does not match the DNS server hostname or if an alias is used for the address, the system cannot resolve the name and its destination, and the VPIM server may reject the message.	String (0–15 characters) Avoid using special characters such as the asterisk (*), tilde (~), etc. in hostnames.	Blank
Domain Name	The domain name that identifies the local system for VPIM. For example, the following VPIM address "Doe, John <1000@LocalDomain.com." When the system sends a VPIM message, the "From" address contains the VPIM domain. When other VPIM systems sends messages to the local system, the "To" address contains an address with the domain as the VPIM domain of the local system. For more information, see "Voice Profile for Internet Mail (VPIM) Networking" on **page 851**. **Note:** To receive and send e-mail messages using VPIM, the Base Server Hostname must be the same as the DNS hostname programmed in the Domain Name field. If the hostname does not match the DNS server hostname or if an alias is used for the address, the system cannot resolve the name and its destination, and the VPIM server may reject the message.	String (0–15 characters) Avoid using special characters such as the asterisk (*), tilde (~), etc. in hostnames.	Blank
DNS Server Primary IP Address	Specifies the DNS Primary IP address for both the Base and Processing Servers that are on the same subnet. However, if the Processing Server requires a different DNS Primary IP Address, this may be programmed in the Advanced IP Settings subfolder. Displayed with a red "X" if DHCP is enabled.	0.0.0.0 – 255.255.255.255	0.0.0.0

Table 95: General IP Settings Fields (continued)

FIELD	DESCRIPTION	RANGE	DEFAULT
DNS Server Secondary IP Address	Specifies the DNS server secondary IP address for both the Base and Processing Servers that are on the same subnet. However, if the Processing Server requires a different DNS Secondary IP address, this may be programmed in the Advanced IP Settings subfolder (see **page 614**). Displayed with a red "X" if DHCP is enabled.	0.0.0.0 – 255.255.255.255	0.0.0.0
DNS Search List	Specifies the DNS Search List for both the Base and Processing Servers that are on the same subnet. However, if the Processing Server requires a different DNS Search List, this may be programmed in the Advanced IP Settings subfolder (see **page 614**). Displayed with a red "X" if DHCP is enabled.	String, up to 200 characters	Blank
Listening Port (Secured)	Specifies the port that the system uses to process secured connections.	1–65535	44000
Listening Port (Unsecured)	Specifies the port number that the system uses to process incoming socket requests (for example, for voice mail, OAI, and so forth). **Note:** The unsecured ports that are used by earlier versions are no longer used for most communications (Web page, DB Programming, Message Print, SMDR, and System Monitor). The unsecured ports are used only for third-party applications, making use of system level or desktop OAI.	none	4000
Listening Port (Unsecured) Enabled	Indicates whether the unsecured listening port is enabled for use. If disabled, applications attempting to use the unsecured port will not be able to connect to the MiVoice Office 250.	Yes or No	Yes
PPP IP Address	Specifies the PPP IP address of the internal modem in the chassis.	0.0.0.0 – 255.255.255.255	192.168.201.202
Enable On-Board TFTP Server	Enables the TFTP Server running on the MiVoice Office 250 that is used for phone upgrades.	Yes or No	Yes

Table 95: General IP Settings Fields (continued)

FIELD	DESCRIPTION	RANGE	DEFAULT
DTMF Decoding Payload	This field specifies the DTMF Payload Type that is to be used fro SIP trunks. Type 101 is becoming more common as the preferred standard for SIP trunk service providers. The payload type is typically negotiated between carriers and devices, but in some cases the SIP trunk service provider does not accept requests. The DTMF Payload Type is made available by the SIP trunk service provider.	96, 101	96
System NAT IP Address	This field specifies the NAT (public) IP address that is used to put: • a SIP gateway behind a NAT. • a SIP peer trunk behind non SIP-aware NAT. The NAT IP address is the address that the system has on the NAT side of the firewall. The range is 0.0.0.0 -255.255.255.255; the default value is 255.255.255.255. *For SIP gateways*, you must also program the SIP Gateway Name field to inform the gateway where the SIP messages are originating from. *For SIP peer trunks*, the SIP Gateway Name is not relevant. For complete information about SIP peer trunks and trunk groups, see "Service Provider SIP Trunks and SIP Trunk Groups" on **page 691**.	0.0.0.0 – 255.255.255.255	255.255.255.255
Maximum Simultaneous Fax Over IP (T.38)	Specifies how many IP networking resources should be reserved/allocated as T.38 and specifies the maximum Fax Over IP calls that can be placed simultaneously. This flag can also be configured in the Resource Reservation Tool. For details, see **page 647**).	0-6	0
Current Expansion Card IP Address	*Read-only.* Displays the current IP address of the Expansion Card on the Base Server. This field applies to systems equipped with a PEC-1 (not shown otherwise).	0.0.0.0 – 255.255.255.255	192.168.200.202
Current Processing Server IP Address	*Read-only.* Displays the current IP address of the Processing Server. This field applies to systems equipped with a PS-1 (not shown otherwise). This value may change if using a DHCP server to obtain an IP address.	0.0.0.0 – 255.255.255.255	192.168.201.201
Current Expansion Card Subnet Mask	*Read-only.* Displays the current subnet mask associated with the Processor Expansion Card on the Base Server. This value should always match the Static Expansion Card Subnet Mask setting.	0.0.0.0 – 255.255.255.255	255.255.255.0

Table 95:   General IP Settings Fields (continued)

FIELD	DESCRIPTION	RANGE	DEFAULT
Current Processing Server Subnet Mask	*Read-only.* Displays the current subnet mask associated with the Processing Server. This value may change if using a DHCP server to obtain an IP address.	0.0.0.0 – 255.255.255.255	255.255.255.0
Current Expansion Card Gateway	*Read-only.* Displays the current gateway IP address associated with the Processor Expansion Card on the Base Server. This value should always match the Static Expansion Card Gateway setting.	0.0.0.0 – 255.255.255.255	192.168.200.1
Current Processing Server Gateway	*Read-only.* Displays the current gateway IP address associated with the Processing Server. This value may change if using a DHCP server to obtain an IP address.	0.0.0.0 – 255.255.255.255	192.168.200.1
Processing Server DHCP Enabled	Determines whether Dynamic Host Configuration Protocol (DHCP) is used for the Processing Server.	Yes/No	Yes
Static Expansion Card IP Address	Specifies the static IP Address for the Expansion Card. This field applies to systems equipped with a PS-1 (not shown otherwise). It appears with a red "X" if DHCP is enabled.	0.0.0.0 – 255.255.255.255	192.168.200.202
Static Processor Server IP Address	Specifies the static IP Address of the Processing Server. This field applies to systems equipped with a PS-1 (not shown otherwise). It appears with a red "X" if DHCP is enabled.	0.0.0.0 – 255.255.255.255	192.168.200.201
Static Expansion Card Subnet Mask	Specifies the static subnet mask associated with the Processor Expansion Card. This setting should be provided by the IP network administrator. Changing this setting resets the IP resource application.	0.0.0.0 – 255.255.255.255	255.255.255.0
Static Processing Server Subnet Mask	This setting is effective and relevant only when DHCP is disabled for the Processing Server. This flag specifies the static subnet mask associated with the Processing Server IP address. This setting should be provided by the IP network administrator. Changing this setting resets the IP resource application.	0.0.0.0 – 255.255.255.255	255.255.255.0
Static Expansion Card Gateway	Specifies the static IP address of the gateway that is used to access the Processor Expansion Card IP port. This setting should be provided by the IP network administrator. Changing this setting resets the IP resource application.	0.0.0.0 – 255.255.255.255	192.168.200.1

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Table 95:   General IP Settings Fields (continued)

FIELD	DESCRIPTION	RANGE	DEFAULT
Static Processing Server Gateway	This setting is effective and relevant only when DHCP is disabled for the Processing Server. This flag specifies the static IP address of the gateway that is used to access the Processing Server IP port. This setting should be provided by the IP network administrator. Changing this setting resets the IP resource application.	0.0.0.0 – 255.255.255.255	192.168.200.1
Processing Server Hostname	Specifies the Processing Server hostname provided by the IP network administrator. This field applies to systems equipped with a PS-1 (not shown otherwise).	String (0–15 characters)	Blank
	Allows you to access the system without having to enter the IP address.	Avoid using special characters such as the asterisk (*), tilde (~), etc. in hostnames.	
	Also used to identify the system on the network.		
	Because the Unified Voice Messaging (UVM) e-mail system requires DNS and a valid hostname, deleting the hostname causes UVM to fail. For more information about UVM, see "Mitel Voice Processing Systems" on **page 836**.		
	Note: To receive and send e-mail messages using VPIM, the Processing Server Hostname must be identical to the DNS hostname programmed in the Domain Name field under System – IP Settings. If the hostname does not match the DNS server hostname or an alias is used for the address, the system cannot resolve the name and its destination, and the VPIM server may reject the message.		

There are warnings associated with IP settings. Below is the explanation of the associated warnings:

- When you access the root folder or the IP Settings folder, if there is a Processor Expansion Card (PEC-1) or Processing Server (PS-1) in the system and DHCP is disabled for two of the three components that exist in the system, Processor Module (PM), PEC-1, and PS-1, and the static subnet mask or gateway settings differ or the static IP addresses appear to be on different subnets, a warning, similar to this, appears: "The Processing Server and Expansion Card appear to be on different subnets. This configuration is not supported." (Only the components that differ are listed.)

- When you disable DHCP for the PM or PS-1 and this results in the condition described just above, the following warning message appears: "When DHCP is disabled, IP addresses on different subnets or gateways are not supported. Please be sure to make additional changes as needed to the static IP settings."

- When you change a static subnet mask or gateway setting, if there is a PEC-1 or PS-1 in the system and the change would make that setting different from either of the others, a warning, similar to this, appears: "When DHCP is disabled, IP addresses on different subnet or gateways are not supported. Click OK to apply this change to the IP settings for the Processor Module, Processing Server, and Expansion Card. (Click Cancel to apply this change only to the Processing Server.)" (Only the components that differ or are being changed are listed.)

  Note that this message appears whether DHCP is disabled or not. Also, if neither a PEC-1 or PS-1 is included in the system, this message does not appear, and behind the scenes, the values associated with those components are automatically changed when the PM value is changed, such that when a PEC-1 or PS-1 is added in, the values are the same. If only a PEC-1 or PS-1 is included, the one not included is changed with selection of OK, or unchanged with selection of Cancel.)

BASE SERVER/PROCESSING SERVER CONNECTION SETTINGS

The Base Server/Processing Server Connection Settings options apply to systems equipped with a PS-1. If your system does not have a PS-1, these options are shown with red Xs.

VIEWING OR CHANGING BASE SERVER/PROCESSING SERVER CONNECTION SETTINGS

You can view or change the settings shown in **Table 96**.

Table 96: Base Server/Processing Server Connection Settings Fields

SETTING	DESCRIPTION	RANGE	DEFAULT
Base Server/ Processing Server Connection Address	Specifies the IP address that the PS-1 uses to connect between the Base Server and Processing Server.	0.0.0.0– 255.255.255.255	192.168.200.201
Base Server/ Processing Server Connection Port	Specifies the port number the PS-1 uses to connect between the Base Server and Processing Server.	1–65535	3400
Base Server /Processing Server Connection Status	For an online session only, this field shows the status of the connection between the PS-1 and Base Server.	Connected/ Disconnected If the connection is set to "Disconnected," you are warned against changing the settings. The warning only appears once during a session.	Disconnected

To view or change Base Server/Processing Server Connection Settings:

1. Select System – **IP Settings**.

2. Double-click **Base Server/Processing Server Connection Settings**.

REFRESHING THE PS-1 TO BASE SERVER CONNECTION STATUS

You can refresh the connection status without closing and reopening DB Programming.

To refresh the connection:

1. Select System – **IP Settings.**

2. Double-click **Base Server/Processing Server Connection Settings**.

3. Right-click **Base Server/Processing Server Connection Status**. The status field is updated.

DHCP SERVER SETTINGS

An embedded DHCP server can issue IP addresses to both Mitel and non-Mitel phones connected to the MiVoice Office 250 when they are brought on-line. The DHCP Server Settings contains four folders, and within each are parameters that need programming.

See the MiVoice Office 250 Database Programming help for details regading specific values and procedures.

Table 97 shows the DHCP Server Settings.

To view or edit DHCP Server Settings:

1. Select **System→IP Settings**.

2. Click **DHCP Server Settings**. The DHCP Server Settings folders appear the the right pane.

Table 97:   DHCP Server Settings Fields

FOLDER NAME	DESCRIPTION
IP Address Ranges	Defines the IP addresses that the DHCP Server will hand out. Each DHCP IP Address Range can be made available to all DHCP clients or to a specific group of clients based on their Vendor Class ID.
Static IP Addresses	Along with the DHCP IP Addresses, defines the IP addresses that the server will hand out. Each DHCP Static IP Address maps to a specific DHCP client.
Options	An Options folder exists in each of the DHCP IP Address Range and DHCP Static IP Address folders. Each DHCP Option belongs to one of the following three scopes: • Global • Specific DHCP IP Address Range • Specific DHCP Static IP Address There are five predefined DHCP Options. Each of the predefined options can be edited or deleted according to the specific needs of the system.
Mitel Tags	Mitel Tag entries provide a way to program parameter tags specific to the Mitel 52xx/53xx IP phones that can be assigned to DHCP Option entries with an option-ID equal to 43 or 125.
DHCP Server Enabled	Specifies whether or not the embedded DHCP Server is enabled.

From the Database Programming menu, select **Operations→DHCP Server Operations**, to perform tasks specific to the DHCP Server.
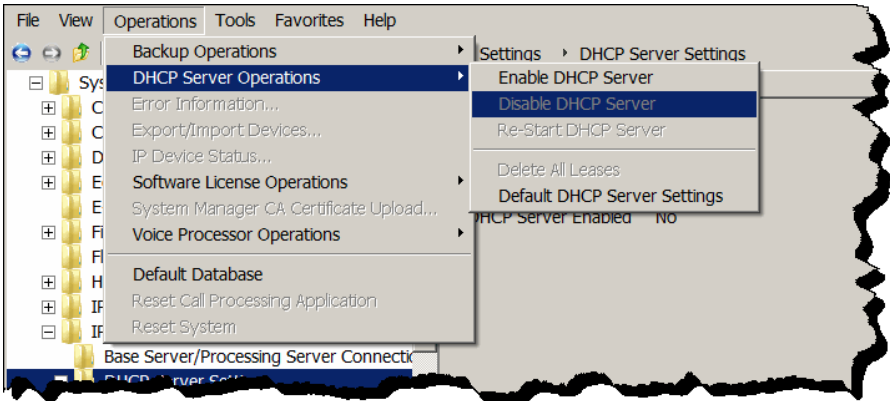
WEB/SSH SETTINGS

Table 98 shows Web/SSH settings.

To view or edit Web/SSH settings:

1. Select System – **IP Settings**.

2. Click **Web/SSH Settings**. Web/SSH settings appear in the right pane.

Table 98:   Web/SSH Settings Fields

FIELD NAME	DESCRIPTION	RANGE	DEFAULT
SSH (Secure Shell) Server Enabled	Specifies whether or not the SSH Server is enabled to log into OLM (On-Line Monitor).	Yes/No	Yes
SSH Server Port	Specifies the port number that the SSH Server uses.	1–65535	22
Web Listening Port (Secured)	Specifies the port that the Web server uses to process secured connections.	1–65535	443
Web Server Enabled	Specifies whether or not Administrative Web Portal (AWP) is enabled.	Yes/No	Yes
Base Server Web Theme	Corresponds to the theme for the AWP. You may set different themes for the Base Server to make it easy to tell which AWP you are viewing based on its theme.	Default, Gray Scale, Mitel	Default
Processing Server Web Theme	Corresponds to the theme for the AWP. You may set different themes for the Processing Server to make it easy to tell which AWP you are viewing based on its theme.	Default, Gray Scale, Mitel	Default

ADVANCED IP SETTINGS

Table 99 shows Advanced IP Settings.

To view or edit Advanced IP settings:

1. Select System – **IP Settings**.

2. Double-click **Advanced IP Settings**. Advanced IP settings appear in the right pane

Table 99:   Advanced IP Settings Fields

FIELD NAME	DESCRIPTION	RANGE	DEFAULT
Current Base Server WINS	(*Read-only.*) Displays the current IP address of the WINS Server for the Base Server. The value may change if using a DHCP server to obtain an IP address.	0.0.0.0 – 255.255.255.255	0.0.0.0
Current Processing Server WINS	(*Read-only.*) Displays the current IP address of the WINS Server for the Processing Server. The value may change if using a DHCP server to obtain an IP address. This field applies to systems equipped with a PS-1. If not equipped, it is not displayed.	0.0.0.0 – 255.255.255.255	0.0.0.0
Static Base Server WINS	Specifies the IP connection's WINS IP address provided by the IP network administrator. Applies to the Base Server.	0.0.0.0 – 255.255.255.255	0.0.0.0
Static Processing Server WINS	Specifies the IP connection's WINS IP address provided by the IP network administrator. Applies to the Processing Server.	0.0.0.0 – 255.255.255.255	0.0.0.0
Processing Server DNS Server Primary IP Address	Used for when the Processing Server needs to point to a different DNS than the Base Server. This field is displayed with a red X if DHCP is enabled. Use case—Set the DNS Server Primary IP Address in the System\IP Settings folder for the Base Server (which affects both the Base and Processing Servers and updates this field as well), and then change this field. Likewise for the following two fields, "Processing Server DNS Server Secondary IP Address" and "Processing Server DNS Search List." This field applies to systems equipped with a PS-1. If not equipped, it is not displayed.	0.0.0.0 – 255.255.255.255	0.0.0.0
Processing Server DNS Server Secondary IP Address	Used for when the Processing Server needs to point to a different DNS than the Base Server. This field applies to systems equipped with a PS-1. If not equipped, it is not displayed. This field is displayed with a red "X" if DHCP is enabled. See Processing Server DNS Server Primary IP Address field on **page 614** for use case.	0.0.0.0 – 255.255.255.255	0.0.0.0

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Table 99: Advanced IP Settings Fields (continued)

FIELD NAME	DESCRIPTION	RANGE	DEFAULT
Processing Server DNS Search List	Used for when the Processing Server needs to point to a different DNS than the Base Server. This field applies to systems equipped with a PS-1. If not equipped, it is not displayed. This field is displayed with a red "X" if DHCP is enabled. See Processing Server DNS Server Primary IP Address field on **page 614** for use case.	0.0.0.0 – 255.255.255.255	0.0.0.0
SIP UDP Listening Port Enable	Enables/disables the external SIP gateway port.	Yes/No	Yes
SIP UDP Listening Port	Specifies the port number that an external SIP gateway uses.	1025–65468	5060

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NTP SERVER CONFIGURATION

By enabling the Network Time Protocol (NTP) support feature, the internal real-time clock (RTC) on the MiVoice Office 250 is updated daily in accordance with an atomic clock that is identified in DB Programming.

With NTP functionality enabled, the MiVoice Office 250 updates the RTC and/or Linux date and time when the following events occur:

- When powered up or rebooted, the system updates both the RTC and Linux date/time using NTP. Otherwise, the system updates the Linux date and time using the RTC.

- Daily at 12:15 AM, the system updates both the RTC and Linux date and time using NTP. Otherwise, the system updates the Linux date and time using the RTC.

- Whenever the NTP, Time Zone, Enable Daylight Savings Time configuration changes within DB Programming, the system updates both the RTC and Linux date and time using NTP. Otherwise, the system updates the Linux date and time using the RTC.

Whenever the Time/Date configuration changes from DB Programming or administrator phone, the system updates both the RTC and Linux date and time with the new date and time. However, if NTP is enabled the new setting is overridden when the next successful NTP synchronization occurs.

This folder contains the fields used to configure the NTP Server. When the Enable Network Time Protocol (NTP) flag is set to No, these fields are displayed with a red "X." Every time this field is changed, the system will attempt an NTP update. The fields can also be programmed in the Configuration Wizard. See "Launching the Configuration Wizard" on **page 59** for instructions on how to launch the wizard. For complete information about the Configuration Wizard, refer to the *MiVoice Office 250 DB Programming Help*.

Table 100:   NTP Server Configuration Fields

FIELD NAME	DESCRIPTION	RANGE	DEFAULT
NTP Server Hostname	Specifies the hostname for the NTP server that is used.	String	north-america.pool.ntp.org (for U.S. systems) or europe.pool.ntp.org (for European systems)
NTP Server IP Address	Specifies the IP address of the NTP Server.	0.0.0.0 – 255.255.255.255	0.0.0.0

NTP SERVER ADVANCED IP SETTINGS

If you are using the NTP server hostname instead of the IP address, you must also configure the following options in "Advanced IP Settings" on :

• Processing Server DNS Server Primary IP Address

• Processing Server DNS Server Secondary IP Address

• Processing Server DNS Search List

NTP SERVER TROUBLESHOOTING

If you have problems with NTP:

• Verify DB Programming.

• Check DNS settings using `ipconfig /all` from the DOS command prompt.

• Check Message Print messages for "NTP Synchronization Successful" or "NTP Synchronization Failed" messages.

• Verify settings and logs on Web page.

LOCAL PROCESSOR MODULE AND EXPANSION CARD IP SETTINGS

NOTICE

System ports used for IP call control and audio may be reserved for other protocols, such as SIP, which uses port 5060 by default. For security purposes, NAT devices, firewalls, or routers occasionally block these ports. If the ports are blocked, you may experience audio or connection problems. If this occurs, change the port numbers in DB Programming to avoid conflicts.

You can program information about each IP connection in the private network. In the IP Connections folder, the right pane displays shortcuts for the Processor module (IP resource settings) and Processor Expansion Card (extra IP resources).

The Processor Module provides IP resource functionality through the IP resource application. See "System and Device IP Settings" on **page 599** for programming information.

The processor module is equipped with a 100-Base T Ethernet port that allows you to connect the system to the LAN.

> **NOTE**   The IP Port does *not* support Simple Network Management Protocol (SNMP).

Before you program the IP port, consult the on-site network administrator for information on any standard practices that are followed, including the use of special characters in hostnames. Providing a static IP address and a hostname is generally considered the "best" practice.

> **NOTE**   Systems equipped with a PS-1 or PEC-1 have two IP addresses. One IP address is for the Processor Module and one IP address is for the Processor Expansion Card. Both IP addresses MUST be on the same subnet mask. This also applies to systems without a PS-1 or PEC-1 that have been upgraded with the Processor Expansion Card. Also, provide a static IP address for the Processor Module. The Processor Expansion Card *must* have a static IP address.

IP RESOURCE PROGRAMMING GUIDELINES

This section describes how to program IP resources for use with IP devices.

NOTICES

- Do *not* connect IP Softphones and other IP phones to the same IP resource. Loss of feature or phone functionality may result.
- Systems equipped with a PS-1 or PEC-1 have two IP addresses. One IP address is for the Processor Module and one IP address is for the Processor Expansion Card. Both IP addresses MUST be on the same subnet mask. This also applies to systems without a PS-1 or PEC-1 that have been upgraded with the Processor Expansion Card. Also, provide a static IP address for the Processor Module. The Processor Expansion Card must have a static IP address.

To program IP resources:

1. In DB Programming (System\Controller), select **Processor Module** or **Processor Expansion Card** (if your system is equipped with a PS-1 or PEC-1). For IP networking, see "Launching the Networking Wizard" on **page 99**.

NOTE	Use a universal numbering plan when programming IP connections. The convention of P6XXY where XX is the node number and Y is the IP resource number is recommended. For example, for the second IP resource on node 3, you would use P6032, and the first IP resource on node 2 would be P6021.

2. Program the IP Settings information (System\Controller\IP Settings), or if you installed the Processor Expansion Card, program the IP Connection information (System\Controller\IP Connection). Use the default port settings when possible. See **page 618** for details.

3. Program voice channels and devices. See "Launching the Configuration Wizard" on **page 59**.

4. Program the devices to use P2P media, if desired.

PROGRAMMING IP RESOURCES AND IP SETTINGS

NOTE	DB Converter provides a Help option with instructions for converting a database.

Depending on the system type you have, the Processor Module (Base Server) appears and provides IP resource configuration options. Or, if the Processor Expansion Card has been installed, the Processor Expansion Card also appears providing additional IP resources.

To view or edit Processor Module or Expansion Card IP Connection settings:

1. Select System – Devices and Feature Codes – IP Connections – Local – <***extension***>.

   If applicable, type the description and user name for the Processor Module or Expansion Card: The description, which appears in all IP connection lists in the database, can consist of up to 20 characters. The username, which appears on display phones, can consist of up to 10 characters. To program the names, select the box and type the entry. Do not use slash (/), backslash (\ ), vertical slash (|), or tilde (~),in usernames. Do not use Control characters in descriptions or usernames.

2. Double-click either **Processor Module** or **Expansion Card** to view IP settings.

   You can program the following IP settings for the local Processor Module and Expansion Card.

   • NAT IP Address on **page 619**
   • Static IP Address on **page 620**
   • Static Subnet Mask on **page 620**
   • Static Gateway on **page 620**
   • Audio RTP Type of Service and Data Type of Service on **page 621**
   • Audio Stream Receive Port on **page 621**
   • IP Terminal TCP Call Control Port on **page 622**
   • IP Terminal General Purpose UDP Port on **page 622**
   • MGCP Receive Port on **page 622**
   • TCP Call Control Port on **page 623**
   • Echo Profile on **page 623**

3. Allocate IP Resources using the Resource Reservations Tool under the Tools menu (see "Resource Reservation Tool" on **page 647**).

4. Configure the Processor Module IP settings, as described on "Local Processor Module and Expansion Card IP Settings" on **page 617**.

5. Configure the IP Connection settings, as described on "Local Processor Module and Expansion Card IP Settings" on **page 617**.

6. Create a universal numbering plan that includes IP connections for forward compatibility. One such plan is to use the convention of "PSnnc" where P is the pause digit, S is the number 6 or 7, nn is the node number, and c is a number between 0–9 identifying the specific IP resource to which the IP connection corresponds.

For troubelshooting information, see **page 623** for Processor Module-related issues and see **page 625** for Processing Server-related issues.

NAT IP ADDRESS

You can program the NAT IP address for the local Processor Module or Expansion Card. A NAT device translates private network IP addresses to one or more public network IP addresses based on NAT translation rules.

NAT devices are installed at the edge of the private network and have internal and external interfaces (and IP addresses). For outgoing IP traffic from the private network to the Internet, NAT translates the source IP address. For incoming IP traffic from the Internet to the private network, NAT translates the destination IP address. NAT devices provide the following advantages:

- Internal IP addresses are hidden from the open Internet and therefore more secure.

- IP addresses are conserved because they are allocated dynamically when needed.

Despite the advantages of NAT devices, they can cause problems for protocols using Peer-to-Peer technologies like multimedia traffic on VoIP networks using SIP.

For more information about NAT, see "NAT Traversal for IP Phones" on **page 113**.

To program the NAT IP Address:

1. Select System – Devices and Feature Codes – IP Connections – **Local**.

2. Double-click either **Processor Module or Expansion Card**.

3. In the **NAT IP Address** option, click the value shown in the **Value** column. The Edit NAT IP Address dialog box appears.

4. Enter the public IP Address that is statically NATed to the private native address. For example, the programmed IP address for the system. The default address is 255.255.255.255.

5. Click **OK**.

STATIC IP ADDRESS

To program the Static IP Address:

1. Enter the default IP address, but do ***not*** include the periods. The default value is *192.168.200.201* and should be changed as appropriate when configuring the network. The Static IP Address field defines the IP connection static IP address provided by the IP network administrator. The system broadcasts changes to an IP connection IP address to the other nodes in the private network as a database update. The IP connection IP address must be kept in sync throughout the network. Conflicting IP addresses can cause the system to behave erratically. The system generates a default value for this field upon creation of the IP connection. To change the IP address, click the field. A dialog box appears.

2. Click **OK**.

> ## NOTICE
>
> Systems equipped with a PS-1 or PEC-1 have two IP addresses. One IP address is for the Processor Module and one IP address is for the Processor Expansion Card. Both IP addresses MUST be on the same subnet mask. This also applies to systems without a PS-1 or PEC-1 that have been upgraded with the Processor Expansion Card. Also, provide a static IP address for the Processor Module. The Processor Expansion Card must have a static IP address.

STATIC SUBNET MASK

To program the Static Subnet Mask:

1. Enter, *without* the periods, the subnet mask associated with the port IP address. By default, this IP address is 255.255.255.0. The Static Subnet Mask field defines the IP connection default network subnet mask provided by the IP network administrator. Changing the default network subnet mask resets the IP resource application.

2. Change the subnet mask:

   a. Click the field. A dialog box appears.

   b. Enter, *without* the periods, the new subnet mask, then click **OK**.

   c. Enter, *without* the periods, the IP address.

   d. Click **OK**.

STATIC GATEWAY

To program the Static Gateway:

1. Enter, *without* the periods, the default IP address of the gateway used to access the port. By default, the address is 192.168.200.1. The Static Gateway field defines the IP connection default IP gateway address provided by the IP network administrator. Changing the default IP gateway address resets the IP resource application.

2. Change the IP gateway address and click the field. A dialog box appears.

3. Click **OK**.

AUDIO RTP TYPE OF SERVICE AND DATA TYPE OF SERVICE

The Audio RTP and Data Types of Service specify the precedence for audio and data packets. The system inserts the values of these two fields into the Type of Service (ToS) field of each audio and data packet. Network devices can then use the value of the ToS field to establish precedence for IP routing. The default value is 0, meaning the packets have no precedence. A value of *184* indicates "IP Precedence." All other values may be used by a network device using Differentiated Services (Diffserv) to establish precedence.

To establish precedence for IP routing:

1. Select System – Devices and Feature Codes – IP Connections – **Local**.

2. Double-click either **Processor Module** or **Expansion Card**.

3. Select either (**Audio RTP**) or (**Data**) **Type of Service**.

4. In the **Value** column, enter the new value in the box.

5. Click out of the field or press **ENTER** to save the change.

AUDIO STREAM RECEIVE PORT

The Audio Stream Receive Port defines the first of the (even-numbered) ports that the system uses to receive audio data packets from IP phones, IP gateways, and remote nodes. The Audio Stream Receive Port must not conflict with port numbers used by other applications running on the system, such as Call Processing, UVM, and so forth.

For example, if the base audio port number is 5000, no other port numbers on the IP connection can fall in the range from 5000 to 5064. Changing the Audio Stream Receive Port resets the IP resources. The system broadcasts changes to an IP connection Audio Stream Receive Port to the other nodes in the private network as a database update. This field corresponds to the off-node IP connection Remote Audio Receive Port (see **"Remote Audio Receive Port"** on **page 627**).

NOTE	Because SIP uses port 5060 (by default) for call control, firewalls, NATs, and routers often treat this port as a "special case" and may block its use by non-SIP packets. If your network currently uses SIP, Mitel strongly recommends that you use a different port range for audio stream ports; otherwise, you may experience audio problems over the IP network.

Although the system allows the Processor Module and Processor Expansion Card port values to overlap, the default database does *not* allow these ports to overlap. **Table 101** shows the value ranges and default values.

Table 101:   Audio Stream Receive Port Ranges and Default Values

MODULE	RANGE	DEFAULT VALUE
Processor Module (PM-1)	1025–64000	6004
Processor Expansion Card (PEC-1)	1025–64000	6604

To change the Remote Audio Receive Port number:

1. Select System – Devices and Feature Codes – IP Connections – **Local**.

2. Double-click either **Processor Module** or **Expansion Card**.

3. Select **Audio Stream Receive Port**.

4. In the **Value** column, type the new number in the box.

5. Click out of the field or press **ENTER** to save the change.

IP TERMINAL TCP CALL CONTROL PORT

The IP Terminal TCP Call Control Port defines the port number the IP resource application uses for call control.

To change the IP Terminal TCP Call Control Port number:

1. Select System – Devices and Feature Codes – IP Connections – **Local**.

2. Double-click either **Processor Module** or **Expansion Card**.

3. Select **IP Terminal TCP Call Control Port**.

4. In the **Value** column, type the new number in the box. The default value varies depending on the phone type.

5. Click out of the field or press **ENTER** to save the change.

IP TERMINAL GENERAL PURPOSE UDP PORT

The IP Terminal General Purpose UDP Port defines the port number the IP resource application uses for general purpose and broadcast messages.

To change the IP Terminal General Purpose UDP Port number:

1. Select System – Devices and Feature Codes – IP Connections – **Local**.

2. Double-click either **Processor Module** or **Expansion Card**.

3. Select **IP Terminal General Purpose UDP Port**.

4. In the **Value** column, type the new number in the box. The range is 1024–65535; the default is 5567.

5. Click out of the field or press **ENTER** to save the change.

MGCP RECEIVE PORT

The MGCP Receive Port defines the port number the MGCP gateway and MGCP phones use for communication.

To change the MGCP Receive Port number:

1. Select System – Devices and Feature Codes – IP Connections – **Local**.

2. Double-click either **Processor Module** or **Expansion Card**.

3. Select **MGCP Receive Port**.

4. In the **Value** column, type the new number in the box. The range is 1024–65535; the default is 2427.

5. Click out of the field or press **ENTER** to save the change.

TCP CALL CONTROL PORT

The TCP Call Control Port defines the port number that off-node IP resources use to connect call control with this IP resource. The TCP Call Control Port cannot conflict with other port numbers on the IP connection. The system broadcasts changes to an IP connection TCP Call Control Port to the other nodes in the private network as a database update. The IP connection TCP Call Control Port must be kept in sync throughout the network. This field corresponds to the off-node IP connection Remote Listening Port. For more information, see **"Remote Listening Port"** on **page 628**.

To change the TCP Call Control Port number:

1. Select System – Devices and Feature Codes – IP Connections – **Local**.

2. Double-click either **Processor Module** or **Expansion Card**.

3. Select **TCP Call Control Port**.

4. In the **Value** column, type the new number in the box. The range of values is 1025–65535. The default is 5570.

5. Click out of the field or press **ENTER** to save the change.

ECHO PROFILE

For information about Echo Profiles, see "Echo Profiles" on **page 735**.

NOTE	If you are using the IP port for the Upload Utility, the Web Listening Port (secured) number must match the port number assigned in the Upload Utility. The default setting for both is 80.

To change the Echo Profile:

1. Select System – Devices and Feature Codes – IP Connections – **Local**.

2. Double-click either **Processor Module** or **Expansion Card**.

3. Select **Echo Profiles**.

4. In the **Value** column, type the new number in the box. The range is 1–65535, and the default is 80.

5. Click out of the field or press **ENTER** to save the change.

TROUBLESHOOTING THE PROCESSOR MODULE (PM-1)

To determine extent of catastrophic hard drive error:

1. Set up an RS-232 connection or keyboard/monitor connection and reset the PS-1.

2. Capture the power-up events and determine if the PS-1 is able boot up.

a. If the PS-1 offers a log-in account, this is not a catastrophic hard drive error. In this situation, Mitel may need remote access to log into the PS-1 to diagnose the problem further.

b. If the PS-1 does not offer a log-in account, this may be a catastrophic hard drive error. Send the power-up events to Mitel before proceeding any further.

3. With assistance from Mitel Technical Support, create a PS-1 recovery CD by using an "ISO" image, or Mitel Technical Support can provide a PS-1 recovery CD.

4. With authorization and guidance from Mitel Technical Support, place the PS-1 recovery CD into the PS-1 optical drive and reset the system.

5. Use an RS-232 connection or keyboard-and-monitor connection to confirm that the installation wiped the previous contents from the PS-1 hard drive and reformatted it based on the CD contents.

Table 102 summarizes the troubleshooting strategies recommended for resolving Processor Module (PM-1) discrepancies.

Table 102:   Processor Module (PM1) Modem Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Users cannot connect to the modem	The Force Minimum Bit Rate flag is enabled, and the connection speed is below the specified minimum.	Disable the Force Minimum Bit Rate flag.
	Poor line quality or other external factors are affecting the connection rate.	Set the Minimum Bit Rate field to 9600 or lower.
An incoming CO call is disconnected when the modem is disabled.	The primary attendant phone is not set.	Make sure the primary attendant phone is set in the Phone-Related Information field so that the call is sent to the attendant when the modem is disabled.

TROUBLESHOOTING THE PROCESSING SERVER (PS-1)

Table 103 summarizes the troubleshooting strategies recommended for resolving PS-1 operational discrepancies.

Table 103:   Processing Server Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
The PS-1 does not connect with the Base Server	Software version mismatch.	Ensure that both the Base Server and the PS-1 are running the same release of v2.x software and that they comply with the requirements identified in the Installation chapter in the *MiVoice Office 250 Installation Manual* .
	The Base Server is not running in Gateway mode.	Use the LCD panel to configure the Base Server in Gateway mode.
	The PS-1 does not have the Base Server connection IP address.	Log into DB Programming on the PS-1 and configure the Base Server connection IP address in *both* of the following folders: • System\IP Settings\Base Server IP Settings • System\IP Settings\Base Server/Processing Server Connection Settings
	The PS-1–Base Server connection IP address does not match the Base Server address.	Verify that the Base Server is using a static IP address. If the Base Server is using DHCP, a new IP address may prevent a successful PS-1–Base Server connection.
	The PS-1 and Base Server do not have the same password and/or port number.	The CP system log file can be used to determine which port the PS-1–Base Server is using for the connection. This information resides in the latest /usr/local/intl/etc/cp/cp_system_log*.* file. Configure the PS-1 password/port numbers in OLM on the PS-1 and the Base Server or in DB Programming.
The PS-1 and Base Server lose communication.	Something prevents the PS-1 and Base Server from communicating with each other.	Ensure that the PS-1 and Base Server are configured on the same switch.
	The PS-1–Base Server connection IP address does not match the Base Server IP address.	Use the PS-1 diagnostics to determine when/if the system started having packet loss. Attempt to associate this with a network impairment issue or a condition where the physical LAN connection was either moved or unplugged.
The PS-1 does not boot up or function due to a catastrophic hard drive error.	The PS-1 hard drive may be severely corrupted.	To determine the extent of the problem, perform the steps on **page 623**.

REMOTE NODE IP CONNECTIONS

You can create remote node IP connections to represent IP resources on other nodes within the network. For each off-node IP connection, the following fields are displayed:

- Node IP Connection Group on **page 627**

- Remote IP Address on **page 627**

- Remote Audio Receive Port on **page 627**

- Remote Listening Port on **page 628**

NOTE	Each node in an IP network must have an off-node IP connection for all other IP nodes in the network. For example, in a three-node network, node 1 must have an off-node IP connection for nodes 2 and 3; node 2 must have off-node IP connections for nodes 1 and 3; and node 3 must have off-node IP connections for nodes 1 and 2. See the example in the figure below. Nodes that do not have the correct off-node IP connections programmed may not be able to communicate with each other.
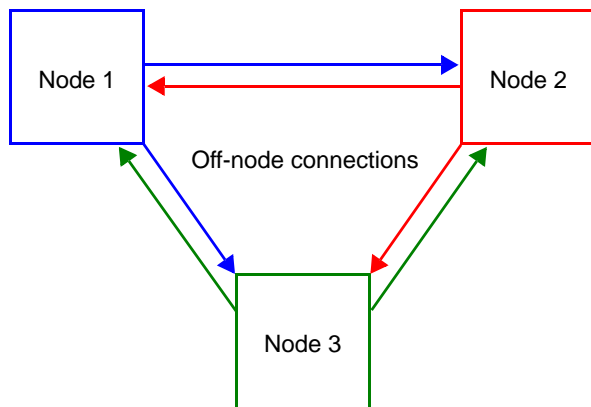
VIEWING OFF-NODE IP CONNECTIONS

You can view off-node IP connections for each remote node.

To view off-node IP connections:

Select System – Devices and Feature Codes – IP Connections – ***<remote node>***. Off-node IP connections appear in the right pane.

CREATING OFF-NODE IP CONNECTIONS

You must create off-node IP connections for each remote IP node in the system network.

NOTE	When you create off-node connections and other IP-related extension numbers, use a numbering plan that associates the extension to the device and the node on which it resides. For example, the first IP resource on node 2 would be P6021 (P6 followed by the node number and then the IP resource number). The second would be P6022, and so forth.

To create an off-node IP connection:

1. Select System – Devices and Feature Codes – IP Connections – *<remote node>* – *<**extension**>*.

2. Right-click in the right pane, and then click **Create Off-Node IP Connection**. The Create Off-Node IP Connection dialog box appears.

3. In the **Starting Extension** list, select an extension that begins with "P6."

4. In the **Number of Extensions** list, select a unique extension for the IP connection.

5. Click **OK**.

NODE IP CONNECTION GROUP

The Node IP Connection Group option links to the node IP connection group in DB Programming to which the IP connection belongs. For more information, see "Node IP Connection Groups for Remote Nodes" on **page 594**.

To view the node IP Connection Group settings:

1. Select System – Devices and Feature Codes – IP Connections – *<remote node>* – *<**extension**>*.

2. Double-click **Node IP Connection Group**.

REMOTE IP ADDRESS

The Remote IP Address defines the remote IP connection IP address provided by the network administrator. This corresponds to the static IP address programmed for the connection on the remote node. For more information, see **"System IP Settings" on page 603**.

To change the Remote IP Address:

1. Select System – Devices and Feature Codes – IP Connections – *<remote node>* – *<extension>* – **Remote IP Address**.

2. In the **Value** column, click the current value. The Edit Remote IP Address dialog box appears.

3. Type the new IP address (do not include the periods). The following IP addresses are not allowed: 0.0.0.0 and 255.255.255.255.

4. Click **OK**.

REMOTE AUDIO RECEIVE PORT

The Remote Audio Receive Port defines the remote IP connection audio receive port. This should match the corresponding IP connection Audio Stream Receive Port (see **page 621**) programmed on the local node.

To change the Remote Audio Receive Port setting:

1. Select System – Devices and Feature Codes – IP Connections – *<remote node>* – *<extension>* – **Remote Audio Receive Port**.

2. In the **Value** column, click the current value, and then type the new port number in the box.

3. Click out of the field or press **ENTER** to save the change.

REMOTE LISTENING PORT
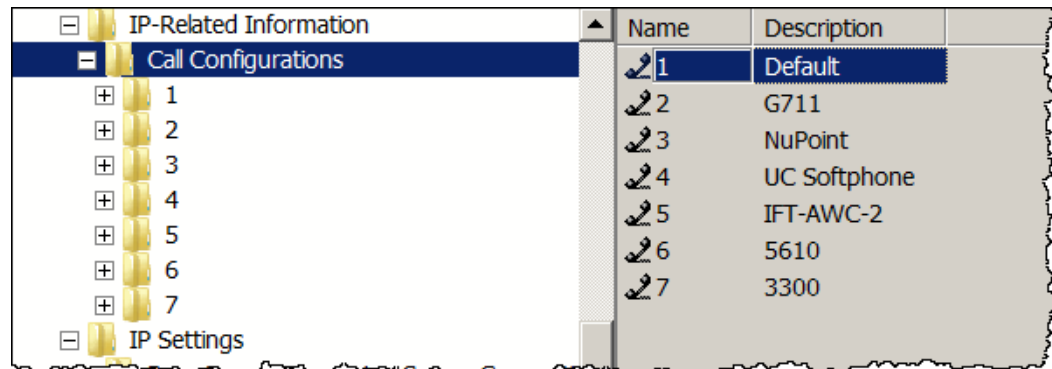
The Remote Listening Port defines the remote IP connection listening port. This should match the corresponding IP connection TPC Call Control Port (see **page 623**) programmed on the local node.

To change the Remote Listening Port setting:

1. Select System – Devices and Feature Codes – IP Connections – *<remote node>* – *<extension>* – **Remote Listening Port**.

2. In the **Value** column, click the current value, and then type the new port number in the box.

3. Click out of the field or press **ENTER** to save the change.

IP CALL CONFIGURATIONS

Call configurations, as shown in the figure below, define the settings that IP phones and gateways use when connected to calls. You can assign multiple devices to a specific call configuration.

By default, all IP devices are placed in Call Configuration 1, which is programmable. You do not need to add SIP phones and MGCP phones to Call Configurations, because these devices negotiate call configurations before establishing a connection. You can program up to 25 different Call Configurations.

The following programmable call configuration names are predefined:

- **Local**: Defines the call configuration settings for IP devices in Local node.

- **Remote**: Defines the call configuration settings for IP devices in Remote node.

- **Bandwidth**: When you create a "Bandwidth" SIP trunk group in System\SIP Peers, the system creates a new SIP trunk group call configuration. By default, the call configuration is named "Bandwidth" and the number is assigned sequentially to the next available number.

- **VoiceFlex**: When you create a "VoiceFlex" SIP trunk group in System\SIP Peers, the system creates a new SIP trunk group call configuration. By default, the call configuration is named "VoiceFlex" and the number is assigned sequentially to the next available number.

- **NetSolutions**: When you create a "NetSolutions" SIP trunk group in System\SIP Peers, the system creates a new SIP trunk group call configuration. By default, the call configuration is named "NetSolutions" and the number is assigned sequentially to the next available number.

- **MiCollab Unified Messaging**: When a SIP voice mail is created in System\SIP Peers, the system creates a new SIP voice mail call configuration. By default, the call configuration is named "MiCollab Unified Messaging" and the number is assigned sequentially to the next available number.

The MiVoice Office 250 supports various SIP Trunk service providers, such as Bandwidth.com, VoiceFlex, NetSolutions, and so on. For an updated list of supported SIP trunk service providers, refer to the Mitel Online Knowledge Base.

NOTE	IP Devices that use P2P to communicate do not have to share the same Call Configuration. When the system detects that IP devices have different settings, it uses the setting that consumes the least amount of bandwidth. You cannot delete the default Call Configuration settings.

To view a list of IP phones that are currently assigned to the call configuration:

1. Select System – IP Related Information – Call Configurations – **Local** (or **Remote**).

2. Double-click **Phones**.

To view IP trunks that are currently assigned to a Call Configuration:

1. Select System – IP Related Information – Call Configurations – **Local** (or **Remote**).

2. Double-click **Trunks**. IP trunks include MGCP gateways and phones and SIP trunks.

This section contains the following information:

*   **Adding Call Configurations** below

*   **Adding IP Phones to the Call Configuration** below

*   Adding Trunks to the Call Configuration on **page 631**

*   Adding SIP Phone Groups to the Call Configuration on **page 631**

*   Adding SIP Trunk Groups to the Call Configuration on **page 631**

*   Adding SIP Voice Mails to the Call Configuration on **page 632**

*   Programming Call Configuration Options on **page 632**

ADDING CALL CONFIGURATIONS

If applicable, add any system trunks or before configuring configurations. See "Creating (Adding) Devices" on **page 355**.

To add a Call Configuration:

1. Select System – IP Related Information – **Call Configurations**.

2. Right-click in the right pane, and then click **Add To Call Configurations List**.

3. Enter the starting ID and the number of call configurations that you want to program.

4. Click **OK**. The new call configurations appear in the list.

5. For each call configuration, select the current description, and then type the new entry in the box. The Call Configuration description can contain up to 20 characters.

6. Click out of the field or press **ENTER** to save the change.

You need to add IP phones to the call configuration as described in the following section.

ADDING IP PHONES TO THE CALL CONFIGURATION

To add IP phones to the list:

1. Select System – IP Related Information – **Call Configurations**.

2. Double-click the call configuration.

3. Double-click **Phones**.

4. In the right pane, right-click and then click **Move To Phones List**.

5. Select the IP phone types, and then click **Next**.

6. Select the devices to add to the list, and then click **Move Items**.

7. Click **Finish**.

ADDING TRUNKS TO THE CALL CONFIGURATION

To add trunks to the list:

1. Select System – IP Related Information – **Call Configurations**.

2. Double-click the call configuration.

3. Double-click **Trunks**.

4. Right-click in the right pane, and then click **Move To IP Trunks List**.

5. Select the trunk type, and then click **Next**.

6. Select the devices to add to the list, and then click **Move Items**.

7. Click **Finish**.

ADDING SIP PHONE GROUPS TO THE CALL CONFIGURATION

To add SIP Phone Groups to the Call Configuration:

1. Select System – IP Related Information – **Call Configurations**.

2. Double-click the call configuration, and then double-click **SIP Phone Groups**.

3. Right-click in the right pane, and then click **Add To SIP Phone Groups List**.

4. Select the SIP Phone Group type, and then click **Next**.

5. Select the devices to add to the list, and then click **Move Items**.

6. Click **Finish**.

ADDING SIP TRUNK GROUPS TO THE CALL CONFIGURATION

To add SIP Trunk Groups to the Call Configuration:

1. Select System – IP Related Information – **Call Configurations**.

2. Double-click the call configuration, and then double-click **SIP Trunk Groups**.

3. Right-click in the right pane, and then click **Move To SIP Trunk Groups List**.

4. Select the SIP Trunk Group type, and then click **Next**.

5. Select the devices to add to the list, and then click **Move Items**.

6. Click **Finish**.

ADDING SIP VOICE MAILS TO THE CALL CONFIGURATION

For MiCollab Unified Messaging voice processing systems only. For more information about MiCollab Unified Messaging, refer to the *MiVoice Office 250 and MiCollab Unified Messaging Integration Guide.*

To add SIP Voice Mails to the Call Configuration:

1. Select System – IP Related Information – **Call Configurations**.

2. Double-click the call configuration, and then double-click **SIP Voice Mails**.

3. Right-click in the right pane, and then click **Move To SIP Voice Mails List**.

4. Select the SIP Voice Mail type, and then click **Next**.

5. Select the devices to add to the list, and then click **Move Items**.

6. Click **Finish**.

PROGRAMMING CALL CONFIGURATION OPTIONS

You can program the following call configuration options:

- **Audio Diagnostics Sampling Period** below

- **Audio Diagnostics Samplings** below

- Audio Frames/IP Packet on **page 633**

- Average In Time Frame Percentage Threshold and Timer on **page 634**

- Minimum Playback Time on **page 635**

- Transmit DTMF Level on **page 635**

- DTMF Encoding Setting on **page 635**

- Speech Encoding Setting on **page 636**

- Fax Control-Messages Redundancy Count on **page 636**

- Fax Page-Data Redundancy Count on **page 637**

- Fax Detection Sensitivity on **page 637**

- Fax Encoding Setting (Fax Transmission) on **page 637**

- Fax Maximum Connection Speed on **page 639**

- Supports RTP Redirect on **page 639**

AUDIO DIAGNOSTICS SAMPLING PERIOD

Mitel IP 5000-series IP phones only. The Audio Diagnostics Sampling Period indicates the time, in seconds, of a sampling interval. 52xx\53xx phones report the statistics at the set interval. Setting the period to 0 seconds sets the diagnostics to report after the audio stream is broken down with an interval of the entire audio stream. Increasing the Audio Diagnostic Sampling Period increases the lost-packet tolerance of phones using the Call Configuration.

If during an interval of the Audio Diagnostic Sampling Period, the received packets percentage drops below the Average In Time Frame Percentage (see **page 634**), the system displays an Insufficient Bandwidth for Voice alarm A032 for the 52xx\53xx phone reporting the bad statistics.

To program the Audio Diagnostics Sampling Period:

1. Select System – IP Related Information – **Call Configurations**.

2. Double-click the call configuration.

3. Select **Audio Diagnostics Sampling Period**.

4. In the **Value** column, select the sampling period (in seconds) from the list. The range is 0–300 seconds; the default value is 5 seconds.

5. Click out of the field or press **ENTER** to save the change.

AUDIO DIAGNOSTICS SAMPLINGS

MiVoice Office 250-series IP phones only. Indicates the number of network statistics samplings that call processing saves. The samplings occur every Audio Diagnostic Sampling Period.

To program the Audio Diagnostics Samplings:

1. Select System – IP Related Information – **Call Configurations**.

2. Double-click the call configuration.

3. Select **Audio Diagnostics Samplings**.

4. In the **Value** column, select the number of samplings from the list. The range is 1–100; the default value is 12.

5. Click out of the field or press **ENTER** to save the change.

AUDIO FRAMES/IP PACKET

The Audio Frames/IP Packet option is the number of audio frames that the system inserts into each packet. The system defines an audio frame as 10 ms of audio. Each audio packet consists of the number of frames set in the Audio Frames/IP Packet option. For example, if you set the Audio Frames/IP Packet option to "2," each audio packet contains two frames (20 ms of audio).

Consider the following when setting this option:

- The lower the value, the lower the latency (delay) in the signal. However, fewer audio frames per packet increases bandwidth consumption (more packets are required), which also increases the chance of jitter and network congestion.

- The higher the value, the higher the latency in the signal. However, more audio frames per packet lowers bandwidth consumption (fewer packets are required), which also decreases the chance of jitter and network congestion.

- Some IP phones support a smaller frames/packet range than this option allows. The system usually tries to negotiate a setting that is within range for both phones. Depending on the call control protocol, this may not always be possible, in which case you must configure the correct value.

To change the number of frames:

1. Select System – IP Related Information – **Call Configurations**.

2. Double-click the call configuration.

3. Select **Audio Frames/IP Packet**.

4. In the **Value** column, select the number audio frames per packet from the list. The range is 1–8 packets. The default value for local phones is 2 packets; the default value for remote phones is 3 packets. For SIP Voice Mails, set this value to **2**.

5. Click out of the field or press **ENTER** to save the change.

AVERAGE IN TIME FRAME PERCENTAGE THRESHOLD AND TIMER

The Average In Time Frame Percentage Threshold / Average In Time Frame Timer settings indicate when network characteristics are inhibiting voice connections.

When the average number of in time frames falls below the Average In Time Frame Percentage Threshold *and* stays below that threshold for the time given by the Average In Time Frame Timer (in seconds), the system displays the INSUFFICIENT BANDWIDTH alarm. For more information about system alarms, see "Alarms" on **page 1127**.

> **NOTE**   If the Average In Time Frame Percentage is set to zero (0), the INSUFFICIENT BANDWIDTH alarm is disabled for all phones in the selected call configuration.

To change the Average In Time Frame Percentage Threshold:

1. Select System – IP-Related Information – **Call Configurations**.

2. Double-click the call configuration.

3. Select **Average In Time Percentage Threshold**.

4. In the **Value** column, select the percentage setting from the list. The range is 0–100%; the default is 60%.

5. Click out of the field or press **ENTER** to save the change.

To change the Average In Time Frame Timer:

1. Select System – IP Related Information – **Call Configurations**.

2. Double-click the call configuration.

3. Select **Average In Time Frame Timer**.

4. In the **Value** column, select the percentage setting from the list. The range is 0–255 seconds; the default is 5 seconds.

5. Click out of the field or press **ENTER** to save the change.

MINIMUM PLAYBACK TIME

The Minimum Playback Time is the time, in milliseconds (ms), that packets wait in the receive buffer before the system plays the audio. The higher this number, the more latency in the signal; however, it is less likely that network problems like jitter would cause lost or late audio packets. The lower the minimum playback time, the less latency there is in the signal; however, there is a greater chance of jitter.

To change the Minimum playback Time setting:

1. Select System – IP Related Information – **Call Configurations**.

2. Double-click the call configuration.

3. Select **Minimum Playback Time**.

4. In the **Value** column, select the time (in ms) from the list. The range is 1–320 ms; the default is 20 ms.

5. Click out of the field or press **ENTER** to save the change.

TRANSMIT DTMF LEVEL

Defines the level at which the system injects tone onto the backplane when receiving an out-of-band DTMF tone. The options are U.S., Japan, U.K., or Mexico. By default, this is the region associated with the language selected.

To change the Transmit DTMF Level country:

1. Select System – IP Related Information – **Call Configurations**.

2. Double-click the call configuration.

3. Select **Transmit DTMF Level**.

4. In the **Value** column, select the country from the list.

5. Click out of the field or press **ENTER** to save the change. The Extended Value and dB columns automatically update to reflect the values associated with the country.

DTMF ENCODING SETTING

The DTMF Encoding Setting is the vocoder type used to send DTMF. The options are G-711 Mu-Law, G-711 A-Law, G-729, and RFC 2833. For MGCP gateways and phones, this value must either match the Speech Encoding Setting field or be set to RFC 2833. By default, this is G.711 Mu-Law [G.711 A-Law in Europe]. If the DTMF Encoding Setting field is set incorrectly, users cannot dial DTMF tones while on a call. For SIP Voice Mails,

To change the DTMF Encoding Setting:

1. Select System – IP Related Information – **Call Configurations**.

2. Double-click the call configuration.

3. Select **DTMF Encoding Setting**.

4. In the **Value** column, select the vocoder type from the list. For SIP Voice Mails, set to **RFC 2833**.

5. Click out of the field or press **ENTER** to save the change.

NOTE	When configured for use with the 5610 Cordless Handset or Unified Communicator SIP Softphone, set the DTMF Encoding Setting in the Call Configuration field to RFC 2833.

SPEECH ENCODING SETTING

The Speech Encoding Setting is the vocoder that the system uses when transmitting speech data. The options are G.711 Mu-Law, G.711 A-Law, G.729, G.729B (VAD), and BroadVoice 32. For MGCP gateways and phones, if the DTMF Encoding Setting is set to RFC 2833, set this value to either G.729 or G.711; if the DTMF Encoding Setting is set to G.729 or G.711, this value must match (for example, set to G.729 or G.711).

To change the Speech Encoding Setting:

1. Select System – IP Related Information – **Call Configurations**.

2. Double-click the call configuration.

3. Select **Speech Encoding Setting**.

4. In the **Value** column, select the vocoder type from the list.

5. Click out of the field or press **ENTER** to save the change.

FAX CONTROL-MESSAGES REDUNDANCY COUNT

Applies to Session Initiation Protocol (SIP) voice mail only. Controls the redundancy count for the fax control messages. The range is 0–7. It is set to 3 by default. The Control-Data redundancy is set to 3 by default because losing even one control message may cause the fax call to fail. Increasing the Control-Data redundancy does not have an impact on the bandwidth. If a red "X" is displayed in this field, you must change the Fax Detection Sensitivity field to any value other than 0.

To program the Fax Control-Messages Redundancy Count:

1. Select System – IP-Related Information – **Call Configurations**.

2. Select the call configuration.

3. Select **Fax Control-Messages Redundancy Count**.

4. In the **Value** column, select or enter the redundancy setting. The range is 0–7; the default is 3.

5. Click out of the field or press **ENTER** to save the change.

FAX PAGE-DATA REDUNDANCY COUNT

Applies to SIP voice mail only. Controls the redundancy count for the fax page data. In general, the more redundancy, the more reliable. However, increasing the Page Data redundancy increases the bandwidth. Losing some Page Data may impact the quality of the image. If a red "X" is displayed in this field, you must change the Fax Detection Sensitivity field to any value other than 0.

To program the Fax Page-Data Redundancy Count:

1. Select System – IP-Related Information – **Call Configurations**.

2. Select the call configuration.

3. Select **Fax Page-Data Redundancy Count**.

4. In the **Value** column, enter the redundancy setting. The range is 0–3. The default is 0.

5. Click out of the field or press **ENTER** to save the change.

FAX DETECTION SENSITIVITY

Applies to SIP voice mail only. Allows you to make false fax detection more or less likely. The higher this number, the less likely the system is to falsely detect fax transmission, but the more likely the system is to fail to correctly detect fax transmission. The lower the number, the more likely the system is to falsely detect fax transmission, but the less likely the system is to fail to correctly detect fax transmission. When this value is zero (0), all faxing options are disabled (indicated with a red "X" next to the options).

NOTE	If you are sending a fax between two MiVoice Office 250s and the Fax Detection Sensitivity settings on the two servers conflict (one is zero and one is nonzero), faxing will not work.

To program the Fax Page-Data Redundancy Count:

1. Select System – IP-Related Information – **Call Configurations**.

2. Select the call configuration.

3. Select **Fax Page-Data Redundancy Count**.

4. In the **Value** column, select or enter the redundancy setting. The range is 0–100; the default is 0, indicating that faxing is disabled. For SIP Voice Mails, set this value to **0**.

5. Click out of the field or press **ENTER** to save the change.

FAX ENCODING SETTING (FAX TRANSMISSION)

Applies to SIP voice mail only. Defines the vocoder that the system uses when the system believes it is transmitting fax data. If a red "X" is displayed in this field, you must change the Fax Detection Sensitivity field to any value other than 0.

The Base Server provides fax over IP (FoIP) capability in Mitel's Private IP Networking environment. FoIP is supported between MiVoice Office 250s in accordance with the ITU-T Recommendation T.38.

Requirements for the FoIP feature include the following hardware and software:

- Two or more MiVoice Office 250s

- Fax machines connected to any CO line

- MiVoice Office 250 Call Processing application

- IP Resource Application

To function correctly, fax machines cannot be connected to IP Single-Line Adapters.

NOTICE

Mitel currently supports T.38 Fax over IP (FoIP) only. If you change any of the fax settings to a value other than the current default setting of T.38, a warning message appears. If you click **OK** when this message is displayed and continue to program the fax settings, you do so at your own risk.

The MiVoice Office 250 supports only a 1-hop T.38 connection, as illustrated in the examples below.
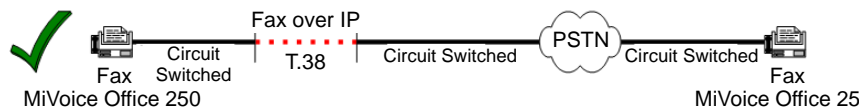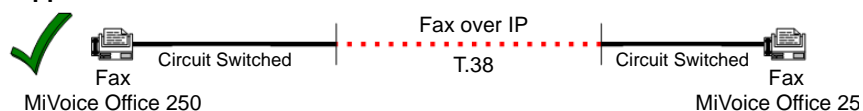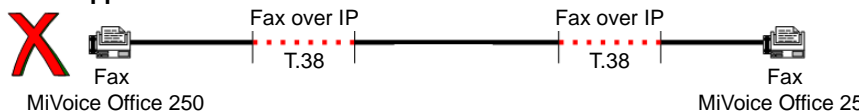
To select the Fax Encoding Setting (Fax Transmission) setting:

1. Select System – IP-Related Information – **Call Configurations**.

2. Select the call configuration.

3. Select **Fax Encoding Setting (Fax Transmission)**.

4. In the **Value** column, select the option from the list. The options are G.711 Mu-Law, G.711 A-Law, and T.38. The default setting is T.38. For SIP Voice Mails, set this value to **G.711 Mu-Law**.

5. Click out of the field or press **ENTER** to save the change.

FAX MAXIMUM CONNECTION SPEED

Applies to SIP voice mail only. Defines the fax connection speed. The available options are 2400 or 4800 or 7200 or 9600 or 12000 or 14400 or No Limit. It is set to *No Limit* by default. If a red "X" is displayed in this field, you must change the Fax Detection Sensitivity field to any value other than 0.

To select the Fax Maximum Connection Speed:

1. Select System – IP-Related Information – **Call Configurations**.

2. Select the call configuration.

3. Select **Fax Maximum Connection Speed**.

4. In the **Value** column, select the option from the list.

5. Click out of the field or press **ENTER** to save the change.

SUPPORTS RTP REDIRECT

Determines whether the SIP peer changes the source port of the RTP and whether the VoIP should start sending its RTP to the new port. It is set to No by default.

To change the RTP Redirect setting:

1. Select System – IP-Related Information – **Call Configurations**.

2. Select the call configuration.

3. Select **Supports RTP Redirect**.

4. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box. For SIP Voice Mails, set this value to **No**.

5. Click out of the field or press **ENTER** to save the change.

SOCKETS

You can set TCP/IP socket connections for the following applications:

- Desktop OAI
- Message Print
- SMDR (for external voice processing systems)
- System Open Architecture Interface (OAI) Level 2

ENABLING OR DISABLING A SOCKET CONNECTION

You can enable or disable socket connections.

To enable or disable a socket connection:

1. Select System – Sockets – <***socket option***>.

2. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

3. Click out of the field or press **ENTER** to save the change.

ENTERING A SOCKET PASSWORD

Socket passwords prevent unauthorized users from accessing socket applications.

To enter a socket password:

1. Select System – Sockets – <***socket option***>.

2. In the Password column, right-click, and then click **Edit Password**. The Edit Password <*option*> dialog box appears.

3. In the **Old Password** box, type the old password, if applicable.

4. In the **New Password** box, type the new password. Typed characters appear as asterisks (***).

5. In the **Confirm Password** box, retype the new password.

6. Click **OK** to exit and save the password. If the entered passwords match, you return to the Password field. If not, you must re-enter the new password.

PHONE AND DEVICE IP SETTINGS

NOTE | Changing IP database settings may drop all calls in progress.

System settings serve as a "template" for IP phones. However, you can change configuration settings for specific IP phones. For example, Dynamic Host Configuration Protocol (DHCP) can be enabled at the system level but disabled for a specific IP phone. DHCP would still be enabled for the other IP phones that are configured to use it at the system level. For more information about supported IP phones, refer to the *MiVoice Office 250 Installation Manual* .

The following sections describe phone IP settings, as shown in:

- Emergency Extensions for IP Devices on **page 642**

- Network Configuration on **page 643**

- Call Configuration on **page 644**

- Reserve IP Resource for Device on **page 645**

- Network Group on **page 645**

- NAT Address Type on **page 645**
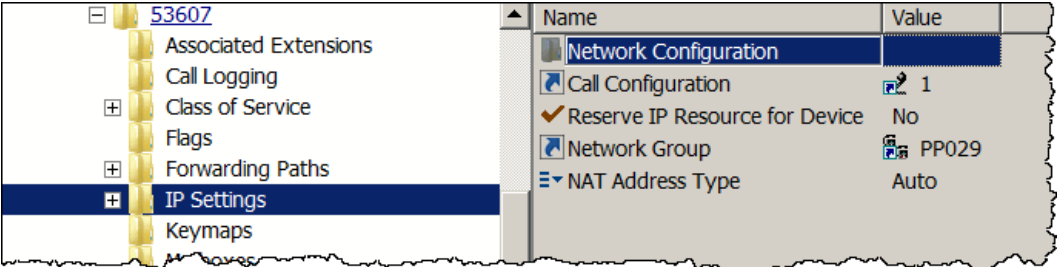
EMERGENCY EXTENSIONS FOR IP DEVICES

The following procedure is recommended programming for a system that has remote loop termination with the need for emergency access for the remote IP phones.

WARNING

Possible Delay in Local Emergency Response to Remote Sites.

You should alert IP and SIP phone users to the following hazardous situations:

- If an Emergency Call phone number is dialed from an IP or SIP phone located at a remote site that is **not** equipped with a correctly configured gateway, the call will be placed from the location where system chassis is installed rather than from the location where the emergency call is made.

   In this situation, emergency responders may be dispatched to the wrong location. To minimize the risk of remote site users misdirecting emergency responders, Mitel recommends regular testing of MGCP/SIP gateway trunk(s) for dial tone.

- If uninterruptible power supply (UPS) protection has not been installed as part of the MiVoice Office 250, IP and SIP phones will **not** operate when electrical power fails either at remote sites or at the main system location.

To place calls during a power failure in this situation, IP and SIP phone users can only use a single line phone connected to one of the power failure bypass circuits built into the system chassis. If a phone connected to a power failure bypass circuit is not available, users should make emergency calls **from a local phone *not* connected to the system**. For details about the Power Failure Bypass feature, refer to the Product Description chapter in the *MiVoice Office 250 Installation Manual* .

Responsibility for Regulatory Compliance:

It is the responsibility of the organization and person(s) performing the installation and maintenance of Mitel Advanced Communications Platforms to know and comply with all regulations required for ensuring Emergency Outgoing Access at the location of both the main system and any remote communication phones. Remote IP and SIP phones may require gateway access to nearby emergency responders.

Emergency Call phone numbers include:

- 911, the default for Mitel systems located in the U.S.

- 999, the default for Mitel systems located in the European market and used primarily in the U.K.

- If applicable, 112, an emergency number used widely in Europe outside of the U.K.

- 112, the default for Mitel systems located in Australia.

- Any emergency number, such as for a police or fire station, that is appropriate for the location of the main system and/or remote phones.

Equipment Damage Hazard. Use only a single appropriate power adapter. Do **not** connect a U.S. power supply and a universal power supply (UPS) to the same device.

Also, when using the UPS with a barrel connector, no devices (for example, hubs) should be inserted between the KS/SLA jack on the adapter and the LAN jack on the phone because power is supplied through the cable.

NOTE	If an installation needs Emergency Outgoing Access across nodes, make sure the Local Trunk Group is the first member in the facility group. This allows cross-node emergency calls to use the Local Trunk Group first and **not** the Remote IP Trunk Group.

The following procedure provides information on how to program an emergency extension for IP devices.

To program an emergency extension for an IP device:

1. Create a Trunk Group that contains the MGCP gateway or SIP trunk(s).

2. Program the Trunk Group as the Emergency Extension for the remote IP phones.

3. Program the Emergency Outgoing Access for the Trunk Group MGCP gateway or SIP trunk(s), give the remote IP phones outgoing access, but remove all other phones and extension lists.

4. Program the Local Trunk Group as the Emergency Extension for local phones.

5. Program the Emergency Outgoing Access for the Local Trunk Group, include all local phones, but *not* the remote IP phones.

6. Create a facility group that contains the Local Trunk Group first, followed by the Trunk Group for the IP phones.

7. Program this facility group in Route Group 1.

8. Program the Emergency Outgoing Access for all node trunk groups to include no devices. Each node should have local trunk termination because emergency outgoing access across nodes is not warranted.

NETWORK CONFIGURATION

The following sections describe network configuration options for 52xx\53xx and 86xx phones.

52XX\53XX PHONE CONFIGURATION OPTIONS

Network Configuration options for 52xx\53xx phones include the following:

- **MAC Address**: The MAC address that was programmed when the device was assigned a circuit. The system reads the MAC address, but it does not set this on the phone.

- **Audio Stream Receive Port**: The 52xx\53xx phone Audio Stream Receive Port has a range of 50098–50508 with a default of 50100, which differs from the other IP phones. This field programs the base receive port number for all channels on this device. Because this device only has one channel, the first audio channel uses base+2 for RTP and base+3 for RTCP, if used.

- Call Control Timeout: The number of seconds that the system waits between heartbeats before it determines that the device is offline.

To view or change 52xx\53xx Phone Network options:

1. Select System – Devices and Feature Codes – Phones – Local – <*extension*>.

2. Double-click **IP Settings**.

3. Double-click **Network Configuration**.

4. Program the desired options, and then click out of each area or press **ENTER** to save your changes.

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IP PHONE NETWORK CONFIGURATION OPTIONS

For descriptions about 86xx phone options, refer to *MiVoice Office 250 Database Programming Help*. You can program the following network settings for IP phones:

> **NOTE** The available fields vary depending on the phone type.

- MAC Address *
- Hostname *
- Static IP Address *
- Static Gateway *
- Static Subnet Mask *
- Static WINS Server *
- IP Address Assignment: BOOTP *
- DHCP Enabled *
- Remote Server IP Address *
- Overwrite Self Programming *
- Telnet Server *
- Web Server *
- Audio RTP Type of Service
- Audio Stream Receive Port
- Password
- Call Control Timeout
- IP Terminal TCP Call Control Port
- IP Terminal General Purpose UDP Port

* The system does not set this information for the 8662, 8622, 8620, and 8600 phones, even though the fields exist in DB Programming. These phones use configuration files over TFTP for configuration or manual programming using Administrative Web Portal (AWP).

CALL CONFIGURATION

The call configuration assigned to the phone. For more information, see "IP Call Configurations" on **page 629**.

To view or change the IP Phone Call Configuration:

1. Select System – Devices and Feature Codes – Phones – Local – *<extension>*.

2. Double-click **IP Settings**.

3. Select **Call Configuration**.

4. In the **Value** column, type the new Call Configuration number in the box.

5. Click out of each area or press **ENTER** to save your change.

RESERVE IP RESOURCE FOR DEVICE

Enables or disable the Reserve IP Resource feature for the phone. For more information, see "Resource Reservation Tool" on **page 647**.

To enable or disable Reserve IP Resource for Device:

1. Select System – Devices and Feature Codes – Phones – Local – <***extension***>.

2. Double-click **IP Settings**.

3. Select **Reserve IP Resource for Device**.

4. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

5. Click out of the field or press **ENTER** to save the change.

NETWORK GROUP

Displays the Network Group to which the device belongs. For more information, see "Network Groups" on **page 588**.

To change the Network Group:

1. Select System – Devices and Feature Codes – Phones – Local – <***extension***>.

2. Double-click **IP Settings**.

3. Select **Network Group**.

4. In the **Value** column, type the new Network Group in the box.

5. Click out of the field or press **ENTER** to save the change.

NAT ADDRESS TYPE

Specifies the NAT address type as follows:

- **Native**: Used for internal IP phones (those that do not pass through near-end NAT).

- **NAT**: Used for external IP phones (those that do pass through near-end NAT).

- **Auto**: Allows the IP phones to be moved inside or outside the firewall (NAT) without programming intervention. For more information, see "Understanding NAT Challenges for SIP Devices" on **page 230**.

> | | The "Auto" option is not applicable to SIP Trunks, MGCP Gateways, MGCP phones, and 52xx\53xx phones. You must manually select the NAT Type setting (Native or NAT).
> | **NOTE** | IP phones retain the NAT Address Type settings after an upgrade. Any database converted to v2.0 or later with the "Auto" option set has this field changed to "Native."
> | | If the MiVoice Office 250 is not connected to a MBG, and there are remote 53xx series phones outside of the network, then the NAT Address Type needs to be set to "NAT".
> | | If the MiVoice Office 250 is connected to a MBG with remote 53xx series phones outside the network, then the NAT Address Type needs to be set to "Native".

For more information about IP phone NAT settings, refer to the *MiVoice Office 250 Installation Manual* .

To change the NAT Address Type:

1. Select System – Devices and Feature Codes – Phones – Local – <***extension***>.

2. Double-click **IP Settings**.

3. Select **NAT Address Type**.

4. In the **Value** column, select the option from the list.

5. Click out of the field or press **ENTER** to save the change.

PROGRAMMING 86XX PHONES IN ITP MODE

When the IP phone is in ITP mode, the system sends down configuration information when the IP phone powers up. However, the only fields that the system can update are as follows:

- MAC Address

- Telnet Server

- Password (changes the password on the 5000 side, *not* the phone side).

- Audio RTP Type of Service

- Audio Stream Receive Port

- Call Control Timeout

- IP Terminal TCP Call Control Port

- IP Terminal General Purpose UDP Port

These options are located under System – Devices and Feature Codes– Phones – <*IP phone*> – IP Settings – Network Configuration. The remaining fields must come from the areas that are programmed in a higher priority configuration source, such as the internal database, self-programming mode, configuration files, and so forth.

RESOURCE RESERVATION TOOL

This section contains the following information:

- **Feature Description** below

- Programming the Resource Reservation Tool on **page 652**

FEATURE DESCRIPTION

> **NOTICE**
>
> **Reservations Reduce Available Shared Resources**. Avoid unnecessarily reserving IP resources. Reserving IP resources reduces the number of resources available for sharing in an oversubscribed system. The more reservations you make, the smaller the pool of shared resources becomes, which increases the potential for delay due to camping on for resources. Reserved resources are used only while the identified function or phone is active. The rest the time, reserved resources cannot be used for other purposes.

You can use the Resource Reservation Tool to reserve IP resources for specific IP phones or system functions. For example, if you reserve two resources for Unified Voice Messaging (UVM), the two resources are removed from the shared pool of resources that is dynamically allocated by the system. The first two simultaneous calls to UVM use the reserved resources. Additional calls to UVM have resources allocated from the shared pool.

After IP resources are reserved for particular purposes, the remaining resources are shared on a first-come, first-served basis.

The following two messages related to reserving resources may appear on display IP phones:

- For calls to UVM, if both reserved UVM resources and shared resources are unavailable, the calling party phone displays the UVM EXT IS BUSY message.

- For calls to a phone, if resources are not available the call is placed in a Camp On state, and the WAITING FOR RESOURCES message displays until resources become available.

RESOURCE RESERVATION CONSTRAINTS

Keep the following constraints in mind when reserving resources for specific phones or functions:

- You should *not* configure reservations unless IP resources are oversubscribed.

- If the system is primarily digital, the demand for IP resources should be minimal. Therefore, IP resources are unlikely to be oversubscribed so reservations should *not* be used.

- Mitel recommends that you reserve IP resources for attendants and other high-traffic users such as call center agents. However, excessive use of reservations degrades the effectiveness of oversubscription by reducing the amount of resources that can be shared. In most cases, only a fraction of the users are likely to be on calls at any given time, which minimizes the likelihood of needing to camp on for IP resources.

- Reserving IP resources for a specific device guarantees that the device can communicate to the system, as long as the IP resources are available when the resource is dedicated.

If the IP resources are not available when requested, the phone camps on until they become available. Reserving IP resources does not guarantee that a call can be completed. For example, if no trunks are available a call could not be completed even if adequate IP resources were available.

- A call to an All-Ring Hunt Group is essentially a call to each hunt group member. The system will consume an IP resource for every IP phone in the All-Ring Hunt Group. Therefore, a single call can consume a great deal of IP resources. Mitel recommends that you minimize the use of All-Ring Hunt Groups in a system that uses oversubscription. If All-Ring Hunt Groups span multiple nodes, the consumption of resources can be even greater.

- A call to a page zone is essentially a call to each member of the page zone. The system consumes an IP resource for every IP phone in the page zone. Therefore, a single call can consume a great deal of IP resources. Mitel recommends that you carefully consider the use of all page zones containing IP phones in a system that uses oversubscription.

- A phone with Background Music enabled is essentially the same as a call to the phone. This is effectively the same as reserving an IP resource for the phone. Therefore, enabling Background Music for a large number of IP phones can consume a great deal of IP resources. Mitel recommends that you minimize the use of background music on IP phones on a system that uses oversubscription. You can also change the feature code for Background Music to restrict use.

FEATURE INTERACTIONS

The system allocates IP resources for each IP device at call setup, and some features deplete the shared pool of IP resources more quickly than others. The following sections discuss what end users can expect to experience with feature interactions.

The following features quickly exhaust IP resources and must be avoided to effectively use oversubscription:

- All-Ring Hunt Groups with IP members or across IP networking

- Large paging zones with IP members or across IP networking

- Reserving a large number of IP resources for T.38 FoIP usage

Simple Call

This interaction describes outgoing and incoming call behavior of a simple call.

- **Outgoing**: An IP phone user wants to call another phone. The IP phone needs to allocate IP resources even if the call will eventually be Peer-to-Peer (P2P). When IP resources are not available, the IP phone camps on for the required IP resources. Once the required IP resources are available, the call proceeds to ring the destination phone.

- **Incoming**: An idle IP phone receives a setup request from a call. Before the setup request is processed, the IP phone needs to allocate IP resources even if the call is P2P. If the required IP resources are not available, the IP phone sends a Camp On signal back to the source phone so it appears the destination IP phone is busy. The IP phone camps on for IP resources and rings once those IP resources become available. If the call was from the CO, the call may ring for a long time if no IP resources become available. However, the user can program a system forward path to voice mail so the call gets routed to the voice

mail after a period of time. If the voice mail is UVM, it is possible to block if the appropriate IP resources are not available.

The vocoder setting in DB Programming under the various call configurations is only a vocoder preference for each device. Some devices that come online may not support the vocoder preference programmed in the database. If the preference is not supported, the phone picks the next best vocoder preference that is supported by that phone. For example, Mitel Audio and Web Conferencing IP phones only support the G.711 vocoder, so the vocoder type that the system will try to allocate will be a G.711 vocoder instead of a G.729 vocoder, which was the programmed vocoder preference for that device.

Agent Help

The request for Agent Help does not complete if the required IP resources are not available. The user is notified by the normal error display AGENT HELP REJECTED and an error tone. The user cannot distinguish between a normal Agent Help failure and an Agent Help failure due to lack of IP resources because the same error message displays for both failures.

Background Music

When an IP phone user wants to listen to Background Music and presses feature code 313 in default mode, the system must allocate required IP resources. If resources are not available, the IP phone camps on for IP resources and the user sees the transient display WAITING FOR RESOURCES on the display phone screen. If the Background Music flag is toggled from DB Programming rather than from an IP phone, WAITING FOR RESOURCES does *not* display on any display phone screen. Background Music begins when IP resources become available. When an IP phone goes offline, IP resources in use for Background Music are relinquished.

NOTE	Mitel recommends avoiding the extensive use of Background Music in oversubscribed IP phone configurations.

Barge-In

To Barge-In on a call, the user must already be silent monitoring a station. The user already has IP resources allocated for the Silent Monitor, so the Barge In request should always succeed. See "Hunt Group Supervisors and Barge-In" on **page 568** for details.

Unified Voice Messaging (UVM)

Any call to the UVM is blocked if sufficient IP resources are not available. If it is undesirable for UVM to block, Mitel recommends that sufficient IP resources be reserved for UVM.

CO Trunk Groups with IP Trunk Members

CO Trunk Groups are basically a type of hunt group and behave almost identically to Simple Hunt Groups. Whenever a call comes into a CO Trunk Group, it tries to hunt for an available trunk to route the call out to the CO. Since MGCP and SIP trunks are CO trunks, they may be in the CO Trunk Group list and also need the appropriate IP resources to be able to make the call. If a CO Trunk Group call reaches an IP trunk and it cannot allocate the required IP resources, it camps on for those resources and changes status to Busy. The CO Trunk Group does not call this IP trunk again until it changes status back to Idle.

Regardless of whether or not the original CO Trunk Group call has hung up or continues to hunt for an available member, any IP phones that could not allocate the appropriate IP resources continue to camp on for those IP resources. Once IP resources become free, the IP phone members who were camping on for IP resources are notified and change their status back to Idle and release the IP resource automatically. If a call is no longer at the CO Trunk Group, nothing happens. If a call is still at the CO Trunk Group, it continues to hunt for an available trunk and can call the IP phone who just changed status to Idle. However, it is possible that when the CO Trunk Group tries to call the IP phone back who just changed its status to Idle, it may not succeed because that IP resource may have been allocated by another IP phone in the meantime.

Conference

Creating a conference or adding a member to a conference behaves similarly to a simple call. See "Simple Call" on **page 648**. Each new IP phone member added to the conference requires sufficient IP resources.

Emergency/911

Mitel recommends that the system be provisioned to have an appropriate number of Emergency/911 IP resources reserved for both IP phones and IP trunks. For instance, if an IP phone makes an Emergency/911 call that must go out an IP trunk, sufficient IP resources for the both the IP phone and IP trunk need to be reserved to ensure the call goes through. If an Emergency/911 call uses ARS to route the call through another node, then the required IP networking IP resources also need to be sufficiently reserved on each node the call passes through to ensure non-blocking. This provisioning includes IP resources for IP trunks if that is the destination.

> ## NOTICE
>
> The system is designed so that an Emergency/911 call has priority over all other IP resource reservations. As long as an IP resource is idle, it will be seized even though not reserved for Emergency/911.
>
> Reserving adequate IP resources does *not* guarantee that an IP phone user will reach an Emergency Responder. The Resource Reservation Tool enables the caller to reach the MiVoice Office 250. Other factors beyond the control of Resource Reservation Tool, such as a trunk not being available, could prevent completion of an Emergency/911 call.

Mitel recommends regular testing of the MGCP/SIP gateway for dial tone and proper connection to local emergency responders upon receipt of a dialed 911 code.

The default value for the number of dedicated Emergency/911 IP resources is 1 and cannot be configured to be 0. When 911 is dialed on the system, an alarm immediately displays on all administrator phones. The alarm displays regardless of whether or not the 911 call camped on or went through and regardless of resources allocated.

Forward and System Forward

If a call is made to an IP phone and no resources are available, the IP phone does not appear as busy. The call to the IP phone will not act upon a manual/system forward no answer/busy if the IP phone cannot allocate the required IP resources. The IP phone camps on for IP resources and rings or redirects the call once the IP resources become available. Manual forwards such as FWD IMMEDIATE work as before.

All-Ring Hunt Groups

With v2.0 or later, the only difference in All-Ring Hunt Groups functionality is that members who cannot allocate sufficient IP resources do not ring, and the hunt group logs their status as Busy until sufficient IP resources become available for those hunt group members to service calls.

Simple Hunt Groups

All IP phone hunt group members are available to their respective hunt groups until the hunt group tries to call them once. At this point, the IP phone rings if it can allocate the appropriate IP resources. If IP resources are not available, the IP phone hunt group member changes status to Busy and camps on for an IP resource. At this point, the hunt group member looks busy to the hunt group (and to all other devices, as well) until the required IP resources are available.

Other phones that have status lamps for this IP phone appear as if the IP phone is busy, even though the IP phone is not on a call. Once a particular hunt group member sends back a Camp On, the hunt group waits until the hunt_member_advance timer goes off before calling the next member.

The scenario described above may be undesirable if the next member could have answered the call immediately. However, because the v2.x hunt group logic works this way Mitel recommends that applications such as Call Center Suite handle the new QU event, which is explained in the preceding OAI section, to be able to differentiate between blocked calls and busy agents.

Regardless of whether or not the original hunt group call has hung up or continues to hunt for an available member, any IP phones that could not allocate the appropriate IP resources continue to camp on for those IP resources.

In summary, one of the following events will occur:

- Once IP resources become available, the IP phone members who were camping on for IP resources are notified and change their status back to Idle and release the IP resource automatically.

- If a call is no longer at the hunt group, nothing happens.

- If a call is still at the hunt group, it continues to hunt for an available member and can call the IP phone that just changed status to Idle. However, it is possible that when the hunt group tries to call the IP phone back that just changed status to Idle, it may not succeed because that IP resource may have been allocated by another IP phone in the meantime.

IP Phone Goes Offline

An IP phone relinquishes any IP resources that are dedicated or in-use for Background Music when it goes offline. If an IP resource is in use for any other reason such as an IP phone having a call on hold when it goes offline, the IP resource is held until either the IP phone comes back online or the IP phone is unequipped from the database. Even if the IP resource is freed, DB Programming continues to count these against the total IP resource count to be able to tell if the system as programmed could potentially block.

PROGRAMMING THE RESOURCE RESERVATION TOOL

This section contains the following information:

- Configuring Resources on **page 652**

- Configuring the Reserved By Function Tab on **page 653**

- Configuring the Reserved By Device Tab on **page 655**

CONFIGURING RESOURCES

This section provides instructions for configuring IP resources.

To configure resources:

1. Make sure you have configured all IP devices (see **page 641**), IP Networking (see **page 642**), UVM ports (see **page 930**), and Call Configurations (see **page 629**).

2. From the Mitel DB Programming Tools menu, select **Resource Reservation Tool**. When the Resource Reservation Tool starts, if the current reservations in the database cause the system to be over-reserved, the following warning appears. To default the reservations, click **OK**, and then click **Save**.
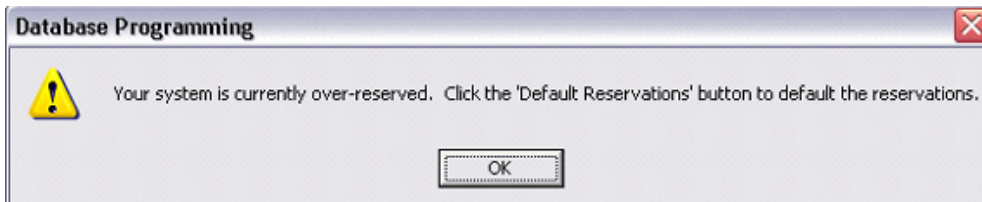
3. Select the **Reserved By Function** tab to reserve resources based on the following three functions. For more information about the Reserved By Function tab, see **page 653**.

    - Emergency/911 resources

    - Unified Voice Messaging port resources

    - Maximum simultaneous Fax over IP (T.38)

4. Select the **Reserved By Device** tab to reserve resources for individual IP devices. All configured IP devices are listed on this tab. For more detailed information about the Reserved by Device tab, see **page 655**.

5. Select the **Advanced** tab to view reservations based on vocoder types for IP Phones, Trunks, or Networking, as well as reservations for Caller ID Transmitters and Caller ID Receivers. Changing the values appearing on this tab is not permitted; the information is displayed for reference only. For more detailed information about the Advanced tab, see **page 655**.

At any time while programming through this tool, you can click one of the following buttons:

- Help**: Opens the online Help.**

- **Default Reservations**: Defaults all the reservations in each tab of the tool to their default values. The default values are not committed to the database until you click the **Save** or **Save and Close** button.

- **Save:** Saves the current reservations of all tabs to the database.

- **Save And Close:** Saves all the current reservations from all tabs and closes the Resource Reservation Tool dialog.

- **Close:** Closes the Resource Reservation Tool without making any changes to the database.

NOTE	If more resources have been reserved than a system can support, the DB Test utility reports an error and defaults the reserved resources fields to 0.

CONFIGURING THE RESERVED BY FUNCTION TAB

The Reserved By Function tab allows you to reserve resources based on functions. The available functions are summarized in Table 104 on **page 654**.
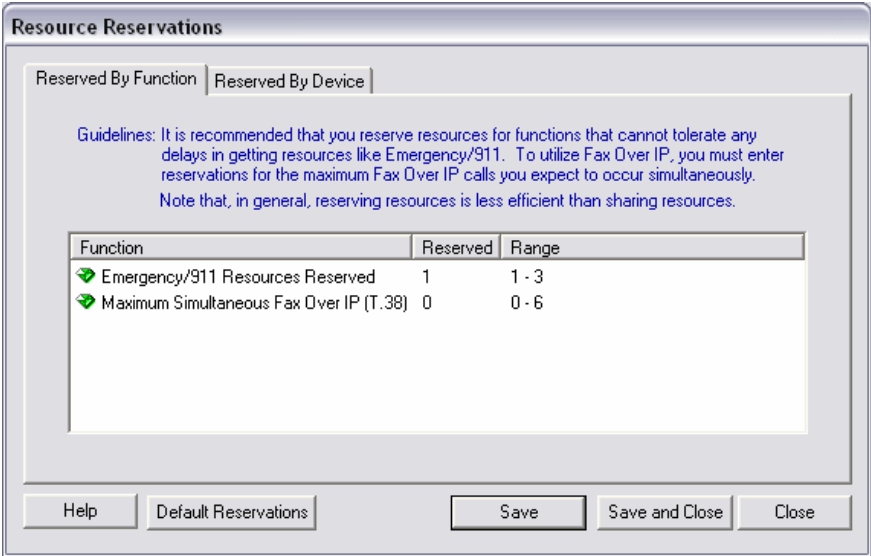
Table 104:   Reserved By Function Fields

FUNCTION TITLE	DESCRIPTION	GUIDELINES	DEFAULT VALUE AND RANGE[1]
Emergency/ 911 Resources Reserved	For each Emergency/911 reservation, the system reserves a G.729 resource. This way, even if all the remaining resources are currently in use, if a user tries to place an emergency call, the call will receive this emergency/911 resource and be able to go through without having to camp on. If more than one emergency call is placed, but only one resource is reserved for emergency, the first call will receive the reserved resource and the other call will have to have a resource allocated from the shared pool. If all shared resources are currently in use, the second call will camp on. Reserving IP resources for a 911 call guarantees that 911 calls can communicate with the MiVoice Office 250. It does *not* guarantee that a call can always be completed (for example, if no trunks are available).	Mitel recommends that the emergency resource reservation count be set at 1 with the understanding that on heavily loaded systems only the first emergency call is guaranteed to have IP resources available. Note that heavily loaded systems will likely have IP resources reserved for specific phones, and that the 911 reservations apply only to the IP phones that do not have IP resources reserved. Mitel also recommends that IP resources be reserved as emergency reservations for IP Gateway trunks used for emergency access. In other words, if IP Gateway Trunks are used for emergency access, the Emergency/911 resource reservation setting should be twice the number of concurrent 911 calls to be supported. For example, if the system uses IP trunks (MGCP or SIP gateway trunks) and uses IP phones, set the Emergency/911 resource reservation value to at least 2: 1 for an IP phone and 1 for the IP trunk.	**Default**: 1 **Range**: 0–3

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Table 104:   Reserved By Function Fields (continued)

FUNCTION TITLE	DESCRIPTION	GUIDELINES	DEFAULT VALUE AND RANGE[1]
Maximum Simultaneous Fax over IP (T.38)	This reservation type is slightly different than the others. It does *not* reserve IP resources from the shared pool. Instead, it reserves T.38 vocoder types for any IP networking call placed. To place a Fax over IP (FoIP) call, a networking resource is allocated (either from the shared pool or from the reservations if any are available). This networking resource must be allocated as a T.38 vocoder, which has a higher "cost" associated with it than the G.711 or G.729. To compensate for the higher cost, each networking reservation utilizes a T.38 resource up to the number specified in this field. Therefore, the maximum number of FoIP calls that can be placed simultaneously will equal the number of reservations made in this field. Because an FoIP call cannot be placed unless there is at least one reservation made, this field also appears in the IP Settings folder. See "System IP Settings" on **page 603**.	Mitel recommends that the number of concurrent T.38 calls be set to a reasonable maximum. T.38 calls are costly in terms of IP resources, and setting this value higher than necessary needlessly takes IP resources away from the shared pool. Note that when reservations are converted from a pre-v4.0 system, existing values are preserved.	**Default**: 0 **Range**: 0–6

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1. The upper limit of the range for each individual reservation can be further limited by the other reservations currently configured

As you change the different reservations, you will see the ranges changing dynamically. A progress bar is displayed as the ranges are calculated along with the message "Updating Ranges." For each given reservation type, you will always know how many additional reservations you can make. Note that the progress bar is just an approximate visual indicator.

To make a reservation:

Click in the Reserved column of the desired reservation type and enter the desired reservation.

CONFIGURING THE RESERVED BY DEVICE TAB

The Reserved By Device tab, as shown in the figure below, allows you to reserve resources for individual IP phones and IP trunks. All IP devices that are currently configured on the system are automatically displayed when you click the Resource Reservation Tool.
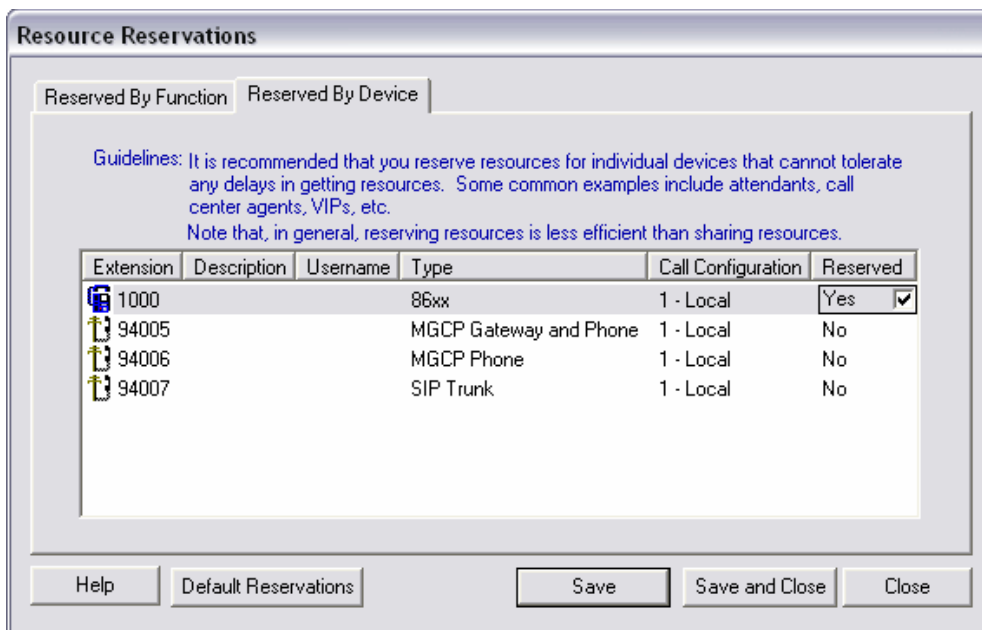
The only configurable option in this screen is the Reserved column. The other options are just provided for reference and may be configured outside of this tool. If desired, click the column headings to sort the list by the entries in that column. To sort the list in reverse alphabetical order, click the column heading once more.

The Reserved column is the same as the Reserve IP Resource for Device flag (see **page 645**). If this flag is enabled, once the reserved device comes online, the system tries to allocate an IP resource for the device. The IP resource comes either from the shared IP resource pool or from the reservations made for the devices by vocoder type in the Advanced tab. If the allocation is successful, the device keeps the resource reserved for itself the whole time it is online, whether it is idle or in use. Note that reserving IP resources for a specific device guarantees only that the device can communicate with the MiVoice Office 250. It does *not* guarantee that a call can be completed, as would occur if no trunks were available.

| **NOTES** | Mitel recommends that you reserve IP resources for attendants and other high-traffic users such as call center agents. However, excessive use of reservations degrades the effectiveness of oversubscription by reducing the amount of resources available to be shared. |
| | One of the important uses of IP gateway trunks is to gain access to Emergency/911 services from remote locations. IP gateway trunks should have IP resources reserved for Emergency/911 access. For details, see Table 104, "Reserved By Function Fields," on **page 10-654**. |

To reserve or unreserve an IP resource for a particular device:

1. Click the row of the desired device and click the Reserved column. A check box appears.

2. Select **Yes** to reserve a resource for this device or select **No** to unreserve the resource.

3. Click **Save**.

TROUBLESHOOTING IP DEVICE ISSUES

This section contains troubleshooting tips for the following issues:

- **IP Devices** below

- IP Device Audio on **page 659**

- IP Device Connection on **page 662**

- IP Device Echo on **page 664**

- IP Device VLAN Tagging on **page 665**

- IP Resource Application (IPRA) on **page 666**

- Oversubscription/IP Resource-Sharing on **page 667**

IP DEVICES

Table 105 summarizes the troubleshooting strategies recommended for resolving discrepancies that may occur when using IP devices. See "System and Device IP Settings" on **page 599** for details.

Table 105: IP Devices Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
IP phone displays VOIP RESOURCE IS UNAVAILABLE	The number of programmed IP devices exceeds the number of available voice channels, and all programmed voice channels are in use.	Verify that IP resources are properly allocated. See the "Resource Reservation Tool" on **page 647**.
Message Print output indicates that a SIP phone extension is invalid or incomplete.	The SIP phone is calling its own number or a number that is programmed but not networked.	Verify the cause but do nothing to attempt to fix the condition.
The Upload Utility is not connecting to the IP devices.	The phone does not have the latest firmware and they are constantly retrying the connection and blocking the Upload Utility.	In general, you should upgrade the IP devices before you upgrade the IPRA. If, however, you have already upgraded the IPRA, you must manually place the devices in the download state before you upload the firmware.
The Upload Utility is not working for IP devices.	You have not entered a password for the connection.	Make sure the Password field is complete before you click **Start**. For more information the Upload Utility password, refer to the Installation chapter in the *MiVoice Office 250 Installation Manual* .
	The IP address entered is wrong.	Reprogram the phone as a static IP address. Or verify you entered the correct IP address.

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Table 105:   IP Devices Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Note: The Upload Utility applies to the 8660 and IP PhonePlus phones only.		
A call was established, but the device cannot send DTMF digits.	The IP phones are programmed for P2P media, but they do not support the same DTMF encoding setting.	Remove the IP devices from the Network Group. Consider adding them to a Network Group that contains IP devices that support the same DTMF encoding setting.
A call was established between two IP phones, but there is no audio.	The IP phones are programmed for P2P media, but they do not support the same vocoder.	Remove the IP devices from the Network Group. Consider adding them to a Network Group that contains IP devices that support the vocoder.
	The IP phones are programmed for P2P media, but they do not support the same number of audio frames per IP packet.	Remove the IP devices from the Network Group. Consider adding them to a Network Group that contains IP devices that support the same number of audio frames per IP packet.
	There is a firewall or NAT between the two devices.	Place the devices into different Network Groups or disable the Use Peer-to-Peer Media flag for the group. See Appendices A and B in the *MiVoice Office 250 Installation Manual* for configuration guidelines.
		Make sure the correct IP ports are open. See the "Phone and Device IP Settings" on **page 641** or the Web interface for the current available ports that can be used.
		Run Network Qualifier to test the ports to make sure they are not blocked. For more information about the Network Qualifier, refer to the *Network Qualifier Reference Manual,* part number 835.2427.
The network is experiencing audio and/or connection problems.	There is a port conflict or the firewall, NAT, or router is blocking the port.	Verify the ports are not blocked. Try changing the ports associated with IP call control and/or audio. Make sure none of them conflict with ports that other protocols use (for example, SIP uses 5060). See Appendices A and B in the *MiVoice Office 250 Installation Manual* for configuration guidelines.

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Table 105:   IP Devices Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Accessing the Record-A-Call or Agent Help feature while using an IP SLA results in the call ringing and then disconnecting as soon as the agent helper or voice mail application answers.	The IP SLA is currently on a peer-to-peer (P2P) call with another IP device. Record-A-Call and Agent Help features are not available on P2P calls.	Remove the IP SLA from a Network Group that supports P2P
Incoming and outgoing calls do not reach the destination.	Incorrect Database or SIP gateway programming. SIP gateway is not on the network. The IPRA is not on the network.	Verify the SIP gateway/IPRA is plugged into the network and the correct IP address is programmed in.
There is a One-Way Audio or No Audio problem.	Incorrect Database or SIP gateway programming, There may be Session Description Protocol (SDP) problems.	Verify that the network groups are programmed correctly for both the SIP trunks and any IP phones involved in the call.
		Identify if the audio should be peer-to-peer or not.

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IP DEVICE AUDIO

Table 106 summarizes the troubleshooting strategies recommended for resolving discrepancies that may occur with IP device audio.

Table 106: IP Device Audio Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Audio quality is poor	The network cannot support VoIP calls. An IP phone does not receive audio packets properly	**Pre-Installation:** Verify the network's ability to support a VoIP call or calls using the Network Test in the Network Qualifier software application. For more information about the Network Qualifier, refer to the *Network Qualifier Reference Manual,* part number 835.2427. **Note:** The Network Qualifier requires a computer on each site. The computers should not be used for any other purpose because other Windows applications could cause jitter. **Post-Installation:** Verify the in-time frames percentage using the In-time Packet Graph in the Receive Audio Status page on the Web interface. You can also watch live values through the Audio Receive Statistics on the Telnet interface. Be sure to watch the percentages on both the IPRA and device. Note that the percentages will be quite low with Voice Activity Detection (VAD) enabled. These low values give some indication of the bandwidth saved by not transmitting silent audio packets.
Audio quality is poor (*Continued*)	Network related issues: there is heavy traffic contending for bandwidth, a hardware problem with switch or router, or defective cabling.	• Increase Audio Frames/IP packet to reduce bandwidth needed; however, increases latency. • Use switches not hubs. • Prioritize the packets through routers; most simply prioritize the audio port (typically UDP 5004), but could also use DiffServ or TOS. • Add more bandwidth. • Paging and background music consume bandwidth; if a site has low bandwidth, they probably should not put all IP phones in the page zone. • Do not use the hub or switch on the IP phone for a Computer or other device that will consume a large amount of bandwidth in bursts or constantly. Run the Network Qualifier to assess your bandwidth. For more information about the Network Qualifier, refer to the *Network Qualifier Reference Manual,* part number 835.2427.

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Table 106:   IP Device Audio Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
An IP phone lost audio suddenly while on a muted call An IP phone lost background music suddenly	IP phones are behind a firewall	When the Voice Activity Detection option is enabled and if any IP phones are behind a firewall, the IP phones may suddenly lose audio (while on a muted call) or lose background music. The IP phone does not send silent audio packets to the IPRA; however, the IPRA continues to send non-silent audio packets to the IP phone. Eventually, most firewalls block this unsolicited IP audio stream from the IPRA to the IP phone. See Appendices A and B in the *MiVoice Office 250 Installation Manual* for configuration guidelines.

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IP DEVICE CONNECTION

Table 107 summarizes the troubleshooting strategies recommended for resolving discrepancies with that may occur when connecting IP devices.

Table 107:   IP Device Connection Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Cannot connect a device to the IPRA	Phone is not programmed	Check the status in the Connected field in the Circuit Status page on the Web interface or the Network Status field in the Network Information on the Telnet.
		If the device is connected, but not operational, then it is a call processing problem. Verify that the System Database is programmed as follows:
		• The MAC address of the device is programmed in DB programming under System – Devices and Feature Codes – *<phones>*
		If the device is not connected, ensure the Device Type, Device ID, and/or Ethernet Address match the device. The following describes where the Ethernet address or device ID is located on each IP device type.
		• IP PhonePlus displays its Ethernet address as it cycles through its power up screens.
		• IP SLA has a sticker with the Ethernet address.
		• IP SoftPhone controls the device ID under the settings option on startup.
		• MGCP Gateway needs the device IP address, instead of the Ethernet address.
		• MGCP phone needs the phone name, instead of the Ethernet address.
		On the Web interface, check the IPRA log to make sure the phone is connecting to the IPRA.

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Table 107: IP Device Connection Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Cannot connect a device to the IPRA	The debug output is not normal	Check the debug output in the Web interface. Normal output appears as follows: <<UDP Broadcast Range/LAN ONLY>> Find server for 00:10:36:00:07:01 from 172.16.10.171:5567 -- port 13 <<UDP PART>> IP ID for 00:10:36:00:07:01 from 172.16.10.171:5567 -- port 13 <<TCP PART>> Connection from 172.16.10.171 on port 13 If the output is not normal, contact Mitel Technical Support.
	The phone does not have the latest firmware	Verify the phone has the latest firmware by viewing the latest version in the phone Web interface. For more information, refer to the Installation chapter in the *MiVoice Office 250 Installation Manual*
General connection issues	The Link LEDs are not lit	Verify the following settings: • The Link LED is lit on the processor module and/or device. • The Link LED is lit on the switch or hub to which these devices connect. • The IPRA or device has a unique IP address. • You can ping the IP address from another computer on the same subnet. • You can ping the IP address of the problem device, and also you can ping in the other direction from the problem device or IPRA. • Appropriate ports for IP devices are opened on a firewall, router, and/or NAT. See Appendices A and B in the *MiVoice Office 250 Installation Manual* for configuration guidelines. To verify the port connection, use the Port Test in the Network Qualifier software application. For more information about the Network Qualifier, refer to the *Network Qualifier Reference Manual,* part number 835.2427.

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IP DEVICE ECHO

Table 108 summarizes the troubleshooting strategies recommended for resolving discrepancies that may occur with IP device echo.

Table 108: IP Device Echo Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
IP phone users hear echo on their phone while talking to an analog trunk	Hybrid balance is not on the correct setting	To reduce echo, first test echo and verify the following settings. Then, follow the instructions on the following page.
		Note: When testing echo, make sure you dial the same phone number using the same trunk. Seize the trunk directly, and do not use the trunk group or ARS. Different trunks have different characteristics.
		Verify the hybrid balance setting. If the trunk does not connect to a public CO, or if the CO is relatively close, then the hybrid balance should be set to "Short." No matter what the current setting is, try the other setting and dial the same number through the same trunk. One setting should be dramatically worse than the other. You should disable the Echo Suppression option so you can actually hear the echo from the beginning of the call.

Note: Refer to the Echo Troubleshooting Guide you can download from the Mitel eDocs Web site (http://edocs.mitel.com).

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
IP phone users hear echo on their phone while talking to *an analog circuit* like on the onboard single lines or trunk	The audio volume for the IP device is too high	To reduce echo from an analog circuit, reduce the audio volume the IPRA drives on to the backplane. This will help the echo canceller adapt quicker. Follow the instructions below:
		1. Adjust the Backplane Transmit Signal Gain option in DB programming. The default setting is 0dB (without any reduction). Any change you make will reduce the volume.
		2. After the adjustment, dial the same number through the same trunk. You should disable the Echo Suppression option so you can actually hear the echo from the beginning of the call. The side effect is that the person on the other end of the call may have a hard time hearing the IP user because the volume is reduced.

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Table 108: IP Device Echo Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
IP users hear a low-volume, clean, clear echo during the beginning of the audio session	Echo Suppression Sensitivity Level is not balanced	Enable the Echo Suppression option. With this option enabled, adjust the Echo Suppression Sensitivity Level during the beginning of an audio session to find the balance between the IP user hearing a slight echo and hearing a half-duplex condition on the handset. The Echo Suppression Sensitivity Level does nothing if the Echo Suppression is disabled. Disable the Echo Suppression when using the IP SLA for fax operations or as a last resort to eliminate the half-duplex condition on the handset during the beginning of the audio session.
An IP user hears raspy or distorted echo as he speaks quite loudly or holds the handset close to his mouth	The Echo Saturation Blocker option is not enabled	Enable the Echo Saturation Blocker option.
An IP user hears choppiness or a half-duplex condition on the handset as they speak quite loudly or holds the handset close to their mouth.	The Echo Saturation Blocker option is enabled	Disable the Echo Saturation Blocker option.

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IP DEVICE VLAN TAGGING

Table 109 summarizes the troubleshooting strategies recommended for resolving discrepancies that may occur with IP device Virtual Local Area Network (VLAN) tagging.

Table 109: IP Device VLAN Tagging-Related Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
The user cannot connect to the Web interface for a multi-protocol phone.	The computer that the user uses may not be in the same VLAN group as the phone.	Make sure that the computer and phone are in the same VLAN group. If not, set the phone VLAN ID to match the computer or disable the VLAN feature for the phone.
The user cannot connect to the 8690 Web interface but could connect to the 8622 or 8662 Web interface. The VLAN ID of the phone port is disabled but the VLAN ID of the downlink port is enabled (not zero).	The VLAN ID of the phone is disabled.	This is a limitation of the 8690 internal phone Ethernet switch. The 8690 Ethernet switch inserts the default VLAN ID which is 1 if the frames from the phone are untagged. Connect the computer to one of the downlink ports which has VLAN ID set to zero.

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Table 109:   IP Device VLAN Tagging-Related Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
DAISY CHAINING IP PHONES IS NOT SUPPORTED BY MITEL Two phones are daisy chained. The phones at the end of the chain cannot connect to the server.	The VLAN ID programmed in the phone at the front of the chain may be programmed with a VLAN ID that does not match the VLAN ID of the phone at the end of the chain.	Program the VLAN ID at the downlink port at the front of the chain to match the VLAN ID of the phone at the end of the chain.
The user powers up the phone with the correct VLAN ID, but the phone receives the wrong IP settings from the wrong DHCP from another VLAN.	The network switch may not support VLAN or wrong VLAN ID is programmed at the switch port of the core switch network.	Make sure that the core network switch is programmed correctly.
The user powers up 8690 but the network settings still show the settings with the old VLAN values.	The phone application may not be updated so the new VLAN settings may not have been propagated to the VPS. If the phone application is up to date, then the new VLAN IDs may not have been sent down to the VPS in time.	Make sure that the phone application is up to date, and reset the phone after the VLAN ID in the phone port has been changed from the networking control panel of Windows CE.

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IP RESOURCE APPLICATION (IPRA)

Table 110 summarizes the troubleshooting strategies recommended for resolving discrepancies occurring with the IPRA.

Table 110:   IPRA Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
IP phones will not come online.	The system pre-allocates one IP resource for each IP device. If the system has insufficient IP resources, the IP device will not be functional.	Get more IP resources. Dedicate IP resources for phone usage.
DESTINATION UNREACHABLE is shown on a phone when making a call across private networking.	There is a blocking condition somewhere in the call path. Blocking conditions may include: • An IP private networking connection with busy IP resources. • A PRI private networking connection with busy B-channels. • A private networking node that is not operational.	Try to place the call again when resources become available. If the condition persists, try the following: • Verify that all nodes in the call path are operational. • Get more IP resources in the case of a busy IP private networking connection. • Program more B-channels in the case of a busy PRI private networking connection.

OVERSUBSCRIPTION/IP RESOURCE-SHARING

Table 111 summarizes the troubleshooting strategies recommended for resolving oversubscription, or IP resource-sharing discrepancies. For additional information, refer to the Oversubscription section in "Appendix E: Engineering Guidelines" in the *MiVoice Office 250 Installation Manual* .

| NOTE | Each IP resource reserved for a particular device or function reduces the amount of resources that can be effectively oversubscribed. |

Table 111:   Oversubscription/IP Resource-Sharing Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
User cannot make calls from their IP phone.	The system attempts to allocate IP resources for each particular call. If IP resources are unavailable, the call will not be completed (including peer-to-peer).	Reserve more IP resources for IP phone usage or dedicate IP resources for a particular IP phone.
User cannot make IP private networking calls.	The system allocates IP resources for each IP private networking call. If no IP resources are available, the call will not be completed.	Reserve more IP resources for IP private networking usage.
IP phone user cannot listen to background music.	Each user must have an allocated IP resource to be able to listen to background music. If no IP resources are available, the user will not hear background music but will camp on for the IP resources.	Reserve more IP resources for IP phone usage or dedicate IP resources for a particular IP phone.
IP phone user cannot hear a page.	If an IP phone cannot acquire the IP resources required for a page, the user will not hear the page.	Reserve more IP resources for IP phone usage or dedicate IP resources for a particular IP phone.
IP phone user does not receive any calls even though their phone is not busy.	In order for an IP phone to ring, it must first be allocated the required IP resources for the call. If no resources are available, the IP phone camps on and does not ring until resources become available.	Reserve more IP resources for IP phone usage or dedicate IP resources for a particular IP phone.
User cannot make IP networking calls even when IP private networking resources are available.	The number of networking calls is limited by the number of IP private networking channels in the license.	Upgrade the IP private networking portion of the license.

- **Reserved By Device**: Indicates how many devices from the Configured column have reservations currently associated with them, as indicated from the Reserved By Device tab.

  For example, once all of the currently configured G.711 phones are determined, this column further investigates which of these devices has the Reserved column set to **Yes** in the Reserved By Device tab. This summary provides a quick reference as to how many IP devices already have reservations made for them.

- **Shared**: Indicates how many IP devices are sharing resources by oversubscription. The number displayed in this column equals the number of Configured devices of this type minus the number of resource reservations indicated in the Reserved column for the type. The number of shared IP devices equals the number of unreserved devices on the system.

Shared Resource Summary Bar: The Shared Resource Summary: bar displays the potential for Camp On. All of the IP devices sharing resources must compete for resources that have *not* been reserved. The potential for Camp On is determined by assuming all of the IP devices sharing resources are in use simultaneously. If the shared pool of resources cannot support the resource requests of all active IP devices, some of the devices camp on until more resources become available. The likelihood of any of these unreserved devices experiencing Camp On is shown in the Shared Resource Summary: bar.

Table 112 summarizes available resource types.

Table 112:   Resource Types[1]

RESOURCE TYPE	DESCRIPTION	GUIDELINES	DEFAULT VALUE AND RANGE[2]
G.711 Phones G.711 Trunks G.711 Networking G.729 Phones G.729 Trunks G.729 Networking	Specifies how many G.711 and G.729 IP resources to reserve for IP Phones, IP Trunks, or IP Networking. G.711 is used for all locally connected IP phones (LAN), and G.729 is used for remotely connected IP phones (WAN or Internet).	Mitel recommends *not* reserving IP resources by device type and vocoder because it reduces the overall efficiency of oversubscription.	**Default:** 0 **Range**: 0–250
Caller ID Transmitter Caller ID Receiver	The system uses IP Resources to generate Caller ID information for single line phones. These resources are needed for a very brief time for each call, but Caller ID transmitting or receiving cannot tolerate Camp On delays. Lack of available resources causes Caller ID transmitting or receiving to fail.	Because Caller ID transmitting or receiving is very transient, Caller ID transmitters and receivers are ideal for oversubscription. Mitel recommends that you *not* reserve Caller ID transmitter and receiver resources.	**Default:** 0 **Range**: 0–250

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Table 112:   Resource Types[1] (continued)

RESOURCE TYPE	DESCRIPTION	GUIDELINES	DEFAULT VALUE AND RANGE[2]
G.711 IP Voice Mail	Specifies the allocation of Voice over Internet Protocols (VoIPs) in terms of calls to an IP voice mail. These resources are used for calls made to/from a SIP peer voice mail. Note that T.38 is currently not supported for the IP voice mail VoIP allocation. For complete information about SIP peer voice mails, see "SIP Voice Mails" on **page 711**.	You should rarely use VoIP reservations because the use of VoIP reservations defeats the sharing model of VoIPs. An example of using VoIP reservations would be an extreme case where the call traffic heavily uses VoIPs. Due to this call traffic, incoming calls to an IP voice mail (for example, MiCollab Unified Messaging) must camp-on waiting for a VoIP. Configuring the IP Voice Mail reservation allows the voice mail system to take precedence in the allocation of VoIPs, thus inflicting the camp-on situations to the other IP devices.	**Default**: 0 **Range**: 0-222
G.729 IP Voice Mail			**Default**: 0 **Range**: 0-108

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1. To sort the list by the entries in a column, click the column heading. To sort the list in reverse alphabetical order, click the column heading once more.
2. The upper limit of the range for each individual reservation can be further limited by the other reservations currently configured.

Chapter 11

SIP PEERS

INTRODUCTION

A SIP peer is an entity with which the MiVoice Office 250 communicates using SIP. This chapter provides programming instructions for SIP peer phones, SIP peer trunks that are provided by service providers, and SIP peer voice mails (MiCollab Unified Messaging):

- **SIP Phones and SIP Phone Groups** below

- Service Provider SIP Trunks and SIP Trunk Groups on **page 691**

- SIP Voice Mails on **page 711**

SIP PHONES AND SIP PHONE GROUPS

This section contains the following information:

FEATURE DESCRIPTION

SIP phones can register with the MiVoice Office 250 and act as local extensions in the system. To support this feature, DB Programming introduces a folder named "SIP Phone Groups." A SIP Phone Group contains a common set of properties for registration that can be shared with either a "stand-alone" SIP phone or multiple SIP phones.

When you create SIP phones, they are configured as "stand-alone" phones by default. Even when the phone is in a stand-alone configuration, the phone is automatically associated with a SIP Phone Group for the purpose of SIP Peer field configuration, but that SIP Phone Group can have only one phone. If you want to use an existing SIP Phone Group and share the properties with other SIP phones, you must assign the SIP phone to a SIP Phone Group when it is being created.

The MiVoice Office 250 currently supports the following SIP phones:

- Mitel Phone Manager Softphone - MPM

- Mitel Unified Communicator (UC) Express SIP Softphone—UCX

- Mitel 5610 Cordless Handset and IP DECT Stand—IP DECT 5610

- Mitel 5603, 5604, and 5607

- Mitel UC360 Collaboration Point (In addition to audio calls, beginning with Release 6.1, also supports video calls.)

- UCA SIP Softphone

NOTE	These SIP phones require a Category F license. The IP DECT Stand itself is not licensed. Also note that handsets are associated with a single IP DECT Stand, so they cannot be used across multiple IP DECT Stands.

For information about system OAI commands that are supported by SIP phones, refer to the documentation included with the v10.20 toolkit.

OAI Applications

The Attendant Console application does not support using a SIP phone as its associated extension.

System and Phones Features

Table 113 on **page 674** shows system features that are compatible or not compatible with SIP phones. Table 114 on **page 676** shows which phone features are compatible or not compatible with SIP phones.

NOTE	The MiVoice Office 250 does not control display of SIP phones. Also, there is no display type defined in the MiVoice Office 250 for SIP phones.

Table 113:   System Feature Compatibility Matrix

SYSTEM FEATURE	DESCRIPTION	SUPPORTED BY:		
		MPM	UCX	IP DECT 5610, MITEL 5603/ 5604/5607 UCA SIP SOFTPHONE
Attendant	A SIP phone can be configured as a Phone Attendant, Local Primary Attendant, or Primary Attendant. However, the SIP phone display may not indicate an incoming recall (Intercom/Outside Hold, Transfer, or Conference call recall).	✓	✓	✓
Audio for Calls	A SIP phone can be programmed to have different audio settings for calls that are ringing, on hold, or camp-on to it.	✓	✓	✓
Caller ID	The MiVoice Office 250 uses the Calling Party Name, Calling Party Number, and Emergency Calling Party Number programmed for a SIP phone in a similar fashion as that of any other phone.	✓	✓	✓
Dynamic Extension Express (DEE)	All DEE functionality can be configureed on a SIP phone, including having it as a user's main extension, and pulling back a call that has been answered on a mobile phone. Handoff-Push functionality is not supported.	✓	✓	✓
Extension ID	Extension IDs can be created for SIP phones so that Voice Processor applications can use these Extension IDs for transferring calls to SIP phones in the local node or networked node.	✓	✓	✓
Extension Lists	SIP phones can be included in IP/Digital Telephone lists. SIP phones are included in system automatic extension lists - All Phones (PP051) and All IP/Digital Telephone (PP052).	✓	✓	✓
File-Based Music-On-Hold	File Based MOH can be set for calls ringing, camped-on, or on hold at the SIP phone.	✓	✓	✓
House Phone	Not supported.			
Mailbox	A SIP phone can be programmed to have mailbox(es) and exhibits the same behavior as that of any other phone.	✓	✓	✓
Off-Node Device	SIP phones can be programmed as an off-node Keyset similar to that of any other phone.	✓	✓	✓

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Table 113:   System Feature Compatibility Matrix (continued)

SYSTEM FEATURE	DESCRIPTION	SUPPORTED BY:		
		MPM	UCX	IP DECT 5610, MITEL 5603/5604/5607 UCA SIP SOFTPHONE
Personal Conference Bridge	When a SIP phone is configured as a User's main extension, the SIP phone extension can be used as a Personal Conference Bridge using the Meet-Me Conferencing feature. Use the Configuration Assistant or User Web Portal to manage conferences.	✓		✓
Peer-to-Peer Media	Supported.	✓	✓	✓
Phone Feature Codes	Not supported, except as indicated.	0, 388, 4, 8, 911	0, 4, 8, 911	0, 4, 5, 8, 911. Ad-hoc conference uses 5.
Ring-In Destination	SIP phones can be programmed as the Ring-In destination for Trunk Groups and Call Routing Tables similar to that of any other IP phone.	✓	✓	✓
SIP Peer Features	Supported.	✓	✓	✓
Trunk Access Codes	The only feature codes that are currently supported by SIP phones are: Emergency Call, Outgoing Call, Automatic Route Selection.		✓	✓
User Web Portal	A SIP phone user can now use the UWP to configure their DEE, DND, Manual Forwarding, manage their Meet-Me Conferencing and manage their account settings (including UVM Enhanced Integration e-mail settings.			
Voice Mail	A SIP phone can be configured to have access to a Voice Mail system. It can use Mitel MiCollab Unified Messaging or UVM as the Voice Mail system. However, the MiVoice Office 250 does not provide display menus or softkeys for SIP phones. MiVoice Office 250 supports accepting calls from SIP phones towards Voice Mail applications in similar way as that of any other phones in the system. For UVM Voice Mail systems, the MiVoice Office 250 uses the programmed language at the SIP phone to determine the language heard by the SIP phone user.	✓	✓	✓

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Table 114:   Phone Feature Compatibility Matrix

				IP DECT 5610, MITEL 5603/ 5604/5607 UCA SIP
		SUPPORTED BY:		
PHONE FEATURE	**DESCRIPTION**	**MPM**	**UCX**	**SOFTPHONE**
Account Codes	Supported by the local feature only.	✓*		
Automatic Call Access	Not Supported.			
Background Music	Not Supported.			
Call Logging	**Missed Calls**: Supported by the local feature only.	✓	✓	✓
	Received Calls: Supported by the local feature only.	✓		
	Dialed Calls: Supported by the local feature only.	✓		
Call Screening	A SIP phone can be programmed to have Call Screening the same way as that of any other phone.	✓	✓	✓
Call Waiting (Camp-On)	A call from a SIP phone towards any other device in the system may camp-on if the device status is Busy or Offline and the corresponding DID/E&M Receive Busy Instead of Camp-on flag is disabled.	✓	✓	✓
	The Camp-On flag at the SIP Phone Group determines whether the calls from SIP phones should camp-on in the following scenarios:			
	• If the concurrent call count limit at the SIP Phone Group-level reaches, any subsequent call from or towards any SIP phone in that group would either camp-on or get rejected based on the Camp-Off flag settings. See **page 725** for details about the maximum concurrent call limit.			
	• If a SIP phone could not acquire a Category F Phones license from the system, any calls from or towards that SIP phone would either camp-on or get rejected based on the Camp-On flag settings.			
	Camp-on indications are not supported.			

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Table 114:   Phone Feature Compatibility Matrix (continued)

PHONE FEATURE	DESCRIPTION	MPM	UCX	IP DECT 5610, MITEL 5603/ 5604/5607 UCA SIP SOFTPHONE
			SUPPORTED BY:	
Conference Calls	**Local Feature**: A SIP phone can initiate a local conference call inviting all the held parties on the SIP phone. To the MiVoice Office 250, it would appear as if there are multiple separate calls initiated by the SIP phone, instead of a conference call. The number of parties that can be involved in such a local conference call depends on the capabilities of SIP phone. Also, the maximum number of concurrent call limit of the SIP phone would limit the number of parties that can be part of such a local conference.		✓	✓
	System Feature: Some SIP Phones (5607/5604/5603/ 5610) can initiate an ad-hoc conference or add parties to an existing conference.		✓	✓
Configuration Assistant	A SIP Phone cannot be configured as an administrator so it cannot use Configuration Assistant to change system settings.		✓	✓
	Do-Not-Disturb (DND): Supported.			
	Forwarding: Supported.			
	DEE: Supported.			
	Conferencing: Supported.			
	Administrator Functions: Supported.			
Directory	**Intercom Directory**: Supported by the local feature only.	✓		
	Speed Dial Directory: Supported by the local feature only	✓		
	Feature Code Directory: Not supported.			

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Table 114: Phone Feature Compatibility Matrix (continued)

PHONE FEATURE	DESCRIPTION	MPM	UCX	IP DECT 5610, MITEL 5603/ 5604/5607 UCA SIP SOFTPHONE
				SUPPORTED BY:
Do-Not-Disturb (DND)	**Local Feature**: A SIP phone may have local DND settings and might reject the calls when in DND mode. If a user changes local DND settings on the SIP phone, the MiVoice Office 250 has no way to know about this change. For example, if a user enables DND on a SIP phone by using the local feature on the phone, calls towards the SIP phone might camp-on because the MiVoice Office 250 determines the SIP phone to be busy or not available. While there are calls camped-on or queued for a SIP phone, and a user disables DND by using the local feature on the phone, the MiVoice Office 250 would not be notified about this change and the calls towards the SIP phone would continue to be in camp-on or queued state.	✓	✓	✓
	System Feature: Supported. However, the SIP phone cannot be used to configure DND using the feature code. Use the Configuration Assistant, User Web Portal or an OAI application instead.	✓	✓	✓
DSS/BLF (PKM) Key	A DSS/BLF (PKM) key can be programmed for Mitel phones that points to a SIP phone. The assigned key reflects the status of corresponding SIP phone. However, a DSS/BLF (PKM) key cannot be programmed on a SIP phone.	✓	✓	✓
Dynamic Extension Express (DEE)	All DEE functionality can be configured on a SIP phone, including having it as a user's main extension, and pulling back a call that has been answered on a mobile phone. Handoff-Push functionality is not supported.	✓	✓	✓

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Table 114:   Phone Feature Compatibility Matrix (continued)

PHONE FEATURE	DESCRIPTION	MPM	UCX	IP DECT 5610, MITEL 5603/ 5604/5607 UCA SIP SOFTPHONE
				SUPPORTED BY:
Emergency Calls	The MiVoice Office 250 allows a SIP phone to make emergency calls. The MiVoice Office 250 does not allow normal calls from a SIP phone to go through in the following situations; however, it would allow emergency calls to go through:	✓	✓	✓
	• The maximum concurrent call limit is reached and a SIP phone is set to be in Busy state.			
	• A SIP phone does not have dynamic binding (active SIP registration) with the system and is set to be in Busy or Offline state.			
	• A SIP phone does not have an active license.			
	• A SIP Phone Group is in Out-of-Service (OOS) operating state.			
	Additionally, a SIP phone can be configured to use Emergency VoIP resources for emergency calls. When a SIP phone places an emergency call, system generates a corresponding Emergency Alarm.			
Group Listen	Not supported.			
Handoff	**Handoff-Pull**: Supported. **Handoff-Push**: Not supported	✓	✓	✓
Hold Calls	Supported by the local feature only.	✓		
Hold Recalls	Not supported.			
Hookflash	Supported for the local feature only.			✓

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Table 114:   Phone Feature Compatibility Matrix (continued)

PHONE FEATURE	DESCRIPTION	MPM	UCX	IP DECT 5610, MITEL 5603/5604/5607 UCA SIP SOFTPHONE
		SUPPORTED BY:		
Hunt Groups	**UCD Hunt Group Number**: Even though a SIP phone can be a member of UCD Hunt Group, it cannot be an agent or supervisor of a Hunt Group. It can also not use Remove/Replace or DND feature code to stop receiving Hunt Group calls. Also, there may not be display indications on SIP Phone for Hunt Group calls or recalls.	✓	✓	✓
	ACD Hunt Group Member: Supported. However, the SIP phone cannot be used to login to the ACD hunt group using the feature code. Use an OAI application instead. The application may login/logout a SIP device, cancel wrap up timer, set DND mode, etc. The following limitations apply: **a.** Automatic Connect is not supported for SIP agents. **b.** Camp On Notifications to Members in DND are not supported for SIP phones.	✓		
	Recall Destination: Supported.	✓	✓	✓
	Hunt Group Remove/Replace: Not supported.			
	Announcement Station: Supported.	✓	✓	✓
	Overflow Station: Supported.	✓	✓	✓
	Agent Help: Supported. However, the SIP phone cannot be used to request help. It must use an OAI application instead. Note: there is no information on the SIP device display if the Agent Help is activated or if a supervisor joined the call. Information about unsuccessful attempt to activate Agent Help feature is displayed in the OAI application though. Also, it is not possible to request help in idle mode—the SIP phone may request the help during an active call only. The Agent Help User-Keyed Extension is not supported for SIP phones.	✓		
	Station Monitor: Supported in agent mode. Note that the SIP phone cannot be configured and used as Supervisor endpoint. Any other supervisor phone could monitor the active call on SIP device, record this call, barge in or steal the call. Similar to Agent Help, there is no information about these features on the SIP phone's display. The SIP user hears beeps when his/her call is being monitored.	✓		
	Group Call Pick Up: Not supported.			

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Table 114: Phone Feature Compatibility Matrix (continued)

PHONE FEATURE	DESCRIPTION	MPM	UCX	IP DECT 5610, MITEL 5603/5604/5607 UCA SIP SOFTPHONE
Intercom Calls	**Calls**: SIP phones can make intercom calls, however, the display on SIP phones may not indicate that it is an intercom call (for example, "IC TO …"). Calls towards SIP phones would always ring regardless of the Ring Intercom flags, unless the SIP phone has local features to enable auto answer.	✓	✓	✓
	Ring Intercom Always: Not supported.			
	Handsfree Feature Code: Not supported.			
	Non-Handsfree Dialing (#): Not supported.			
Intelligent Directory Search	Not supported.			
Manual Forwarding	**Local Feature**: SIP phones may have a local forwarding functionality where it allows user to forward calls to any specified destination. The MiVoice Office 250 may not view such calls as forwarded calls.	✓	✓	✓
	System Feature: A SIP phone supports Manual Forwards. The SIP phone cannot be used to configure the Manual Forward using the feature code. Use the Configuration Assistant, User Web Portal or an OAI application instead.	✓	✓	✓

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Table 114: Phone Feature Compatibility Matrix (continued)

PHONE FEATURE	DESCRIPTION	MPM	UCX	IP DECT 5610, MITEL 5603/5604/5607 UCA SIP SOFTPHONE
			SUPPORTED BY:	
Messages	**Station Messages**: Not supported.			
	Alternate Message Source: Not supported.			
	Silent Messages: Not supported.			
	Message Waiting Indication (MWI): The MiVoice Office 250 sends unsolicited MWI notifications towards SIP phones to indicate the current MWI status for that phone. At this point, the MiVoice Office 250 does not support accepting MWI subscriptions from SIP phones. If a SIP phone tries to subscribe for "message-summary" event for MWI updates, the MiVoice Office 250 rejects the SIP SUBSCRIBE request with the "405 - Method Not Supported" response. If the MiVoice Office 250 is using MiCollab Unified Messaging as the Voice Mail system, the unsolicited MWI notifications towards SIP phones would only indicate the presence or absence of Voice Mail message(s). Whereas, if UVM is the Voice Mail system, the unsolicited MWI notifications towards SIP phones would indicate the presence or absence of Voice Mail message(s) as well as the message count.	✓	✓	✓
Microphone Mute	**Local Feature**: A SIP phone may have a local feature to mute/unmute the call. For example, the UCX, IP DECT 5610, and Mitel 5603/5604/5607 have local features to mute/unmute the call.	✓	✓	✓
	System Feature: Not supported.			
Multilingual Capability	SIP phones may have multilingual capability. In addition, an administrator can program the language for SIP phone that the callers would hear for Voice Mail and Configuration Assistant.	✓		
Music-on-Hold (MoH)	MOH can be set to Silence, Tick Tone, Ringback, MiVoice Office 250 or File-Based MOH for SIP phones.	✓	✓	✓
Off-Hook Voice Announce	Not supported.			
On-Hook Monitoring	**Local Feature**: SIP phones may allow on-hook monitoring as a local feature.	✓	✓	✓
	System Feature: Not supported.			
Outside Calls	SIP phones can place and receive outside calls. The displays on SIP phones may not show Mitel MiVoice Office 250 formatted outside number as part of caller ID.	✓	✓	✓

Table 114:   Phone Feature Compatibility Matrix (continued)

PHONE FEATURE	DESCRIPTION	SUPPORTED BY:		
		MPM	UCX	IP DECT 5610, MITEL 5603/5604/5607 UCA SIP SOFTPHONE
Outgoing Access	A SIP phone can be programmed to have outgoing access the same way as that of any other phone.	✓	✓	✓
Outgoing Extension	An Outgoing Extension can be programmed for a SIP phone to have outgoing access the same way as that of any other phone.	✓	✓	✓
Paging	**Page Zones**: A SIP Phone can dial the specific page zone extension to place a page. **Page Calls**: Not supported. **Enable/Disable Paging Feature Code**: Not supported. A SIP Phone cannot dial the page feature code (7). At this point, there is no established mechanism for SIP Phones to use Mitel MiVoice Office 250 specified feature codes.	✓	✓	✓
Phone Feature Codes	Not supported.			
Queue	Not supported.			
Record-A-Call	Not supported.			
Redial Calls	**Local Feature**: SIP phones may provide a redial functionality as a local feature.	✓	✓	✓
	System Feature: Not supported.			
Redirect Calls	**Local Feature**: SIP phones may provide a redirect functionality as a local feature.	✓	✓	✓
	System Feature: No supported.			
Reminder Message	Not supported.			
Speed Dials	**System Speed Dial**: Supported by the local feature only.	✓		
	Station Speed Dial: Supported for the local feature only.	✓	✓	✓
System Forwarding	A SIP phone supports System Forward functionality, except Ring Principle Once. System Forward cannot be enabled or disabled using the SIP phone. This setting must be configured by the administrator using DB Programming.	✓	✓	✓

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Table 114:   Phone Feature Compatibility Matrix (continued)

PHONE FEATURE	DESCRIPTION		SUPPORTED BY:	
		MPM	UCX	IP DECT 5610, MITEL 5603/ 5604/5607 UCA SIP SOFTPHONE
Transfer Calls	A SIP phone can either do a Blind Transfer and/or an Attended Transfer, depending on its capabilities. It can also receive calls transferred by other phones. In other words, in a Transfer Call context, a SIP phone can act as Transferrer, Transferee, or Transfer Target.	✓	✓	✓
	Transferring Conference Calls: Supported for the local feature only.		✓	✓
	Transfer To System Forward: Not supported.			
	Transfer Timers: Supported.	✓	✓	✓
	Transfer-to-Connect: Not supported.			
	Transfer-to Ring: Supported for the local feature only. The calls transferred to a SIP phone would always be Transfer-to-Ring and cannot be Transfer-to-Hold, even though the transferring phone tried the Transfer-to-Hold option.	✓	✓	✓
	Transfer-to-Hold: Not supported. Unless the SIP phone has the capability to initiate a Transfer-to-Hold request, whenever it initiates a Transfer Call (Blind or Attendant), it would always be treated as a Transfer-to-Ring request. It uses the local transfer feature to initiate the transfer.			
	Transfer Recalls: A SIP phone can handle the transfer recall. However, the SIP phone display may not indicate that it is receiving a transfer recall. It would appear as if there is a new incoming call and after the user answers that call, it would reconnect the call that is being transferred.	✓	✓	✓
	Reverse Transfer: A SIP phone user can now perform the reverse transfer feature to answer a call that is ringing or holding at another phone. Note that a call that is ringing or holding at a SIP phone cannot be answered with the reverse transfer feature.	✓	✓	✓

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* Requires Professional or Team Leader License

CREATING SIP PHONES

This section describes how to create SIP phones. You can create up to 250 SIP phones by default.

For additional programming instructions, refer to the *UC Express IT and Administrator Guide* or the *Mitel 5610 IP DECT Stand Configuration and Administration Guide*.

To create a SIP phone:

1. Select System – Devices and Feature Codes – **Phones**.

2. Right-click anywhere in the right pane, and then select **Create SIP Phone**. The Create SIP Phone Extension dialog box appears.

3. Select a starting extension for the phones and the number of extensions.

4. Click **OK**. The system creates a new SIP Phone Group for each of the SIP phones. The SIP phones are created in a stand-alone configuration by default. The associated SIP Phone Group is displayed in System\Devices and Feature Codes\SIP Peers\SIP Phone Groups (see **page 686** for details).
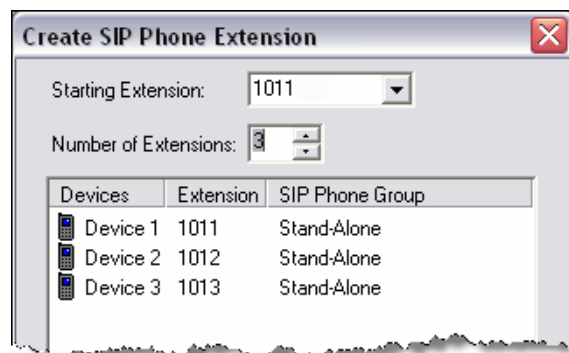
To assign a SIP phone to an existing SIP Phone Group:

1. Right-click the SIP phone in the Create SIP Phone Extension dialog box, and then select **Add to existing SIP Phone Group**.

2. When the Change SIP Phone Group dialog box appears, click **Next**. The existing SIP Phone Groups, similar to the ones below, appear.
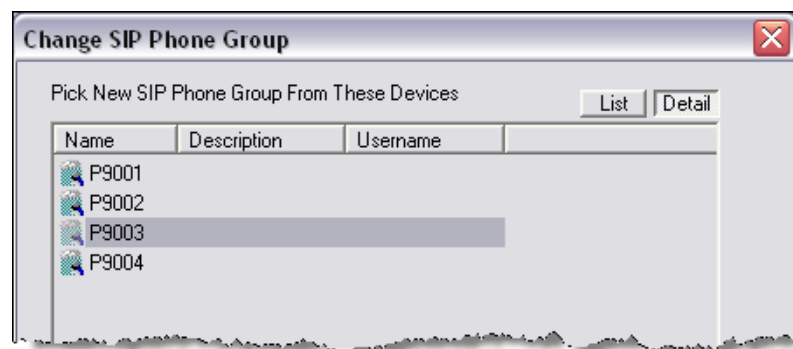
3. Select the SIP Phone Group that you want to use, and then click **Finish**. The new SIP Phone Group appears in the SIP Phone Group column in the Create SIP Phone Extension dialog box.

To switch the SIP Phone Group back to Stand-Alone:

Right-click the device in the Create SIP Phone Extension dialog box, and select **Create new Stand-Alone SIP Phone Group**. The SIP Phone Group column changes to "Stand-Alone" and a new stand-alone SIP Phone Group is created.

PROGRAMMING SIP PHONES

Follow the instructions below to program SIP phones. For information about SIP phone options, see "Programming Phone Options" on **page 391**.

To program SIP phone options:

Select System – Devices and Feature Codes – Phones – Local – *<**SIP phone**>*.

NOTE	The configuration applications for both the 5610 Cordless Handset and UC Express SIP Softphone use the assigned SIP Phone extension number and passcode. If these values do not match, the corresponding SIP phone will not function.

CREATING SIP PHONE GROUPS MANUALLY

When you create stand-alone SIP phones, SIP Phone Groups are automatically created in the SIP Phone Groups folder in System\Devices and Feature Codes\SIP Peers\SIP Phone Groups. This section explains how to create SIP Phone Groups manually. You can create up to 250 SIP Phone Groups by default.

To create SIP Phone Groups manually:

1. Select System – Devices and Feature Codes – SIP Peers – **SIP Phone Groups**.

2. Right-click anywhere in the right pane. The Create SIP Phone Groups Extension dialog box appears.

3. Select **Create SIP Phone Group**. The Create SIP Phone Groups Extension dialog box appears.

4. Select a starting extension for the groups and the number of extensions, and then click **OK**. When you create a SIP Phone Group, the Call Configurations subfolder under System\IP-Related Information creates a new call configuration for the SIP Phone Group by default (see "IP Call Configurations" on **page 629** for details).

5. After you have created a SIP Phone Group, assign SIP phones to the group as described on **page 685**.

PROGRAMMING SIP PHONE GROUPS

Each SIP Phone Group has the Configuration and Phones folders.

CONFIGURING THE SIP PHONE GROUP PROGRAMMING OPTIONS

Table 115 show configuration options that are available for SIP Phone Groups. The SIP Phone Group configuration options are similar to the configuration options for SIP Trunk Groups.
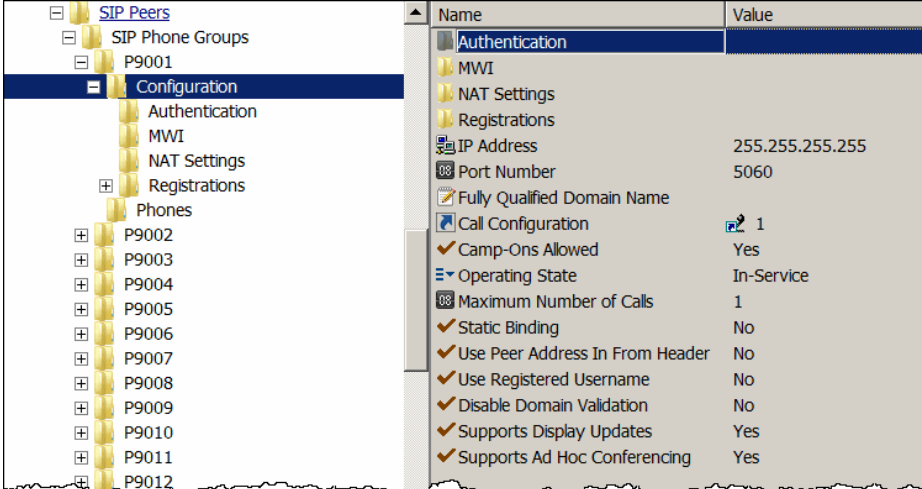
To program SIP phone group programming options:

Select System – Devices and Feature Codes – SIP Peers – SIP Phone Groups – *<SIP phone group>* – **Configuration**.

For detailed programming instructions, see **page 717**.

| NOTE | When configured for use with the 5610 Cordless Handset, set the DTMF Encoding Setting in the Call Configuration field to RFC 2833. |

Table 115: SIP Phone Group Options

VIEWING THE SIP PHONE LIST

You can view the SIP phones that are associated with SIP Phone Groups.

To change the settings of a SIP phone:

Double-click the SIP phone that you want to configure.

SIP VIDEO CALLS

MiVoice Office 250 has no on-board capabilities to support video, or to act as a STUN server. A Peer-to-Peer connection between two SIP endpoints (known as SIP P2P) is the only option for Video calls between endpoints connected to the MiVoice Office 250.

The MiCollab desktop client (Mitel UC360) and 3rd party SIP phones are configured on the MiVoice Office 250 system as SIP endpoints. SIP signaling is used to setup the calls between these types of phones. As a result, these endpoints have the option to escalate an established audio call to a video call.

The following devices support SIP video calls:

- X-Lite

- Bria

- MiVoice Video Phone

- MCD

- MBG

The table below summarizes the supported codecs for these devices.

	X-LITE	**BRIA**	**MIVOICE VIDEO PHONE**
X-Lite	H.263 H.263+ (1998)	H.263 H.263+ (1998)	No common codec. No Video Call
Bria		H.263 H.263+ (1998) H.264 (base profile) VP8	H.264 (base profile)
MiVoice Video Phone			H.264(base profile) H.264(high profile)

SIP P2P video calls support the following features:

- Escalation Video

- Hold/Retrieve

- Forward

- Transfer

- Conference (if the device supports local conferencing)

- Private Networking.
  For example, the SIP device can be on another node. Note that the extension of the Network Groups must match, and the **Peer to Peer Media** flag in Database Programming needs to be enabled on both nodes. Both nodes must be running MiVoice Office Release 6.1 and higher software.

- SIP P2P video functionality can work with MiVoice Business using Private Networking via SIP Peer Trunk Groups.

Only SIP to SIP call scenario will be supported for escalating the audio call to a video call. If any non-SIP devices are involved in the call (TDM/Conference, traditional trunks), then the escalation to a SIP video call is impossible.

The diagram below illustrates the cases in which the SIP video calls will work.
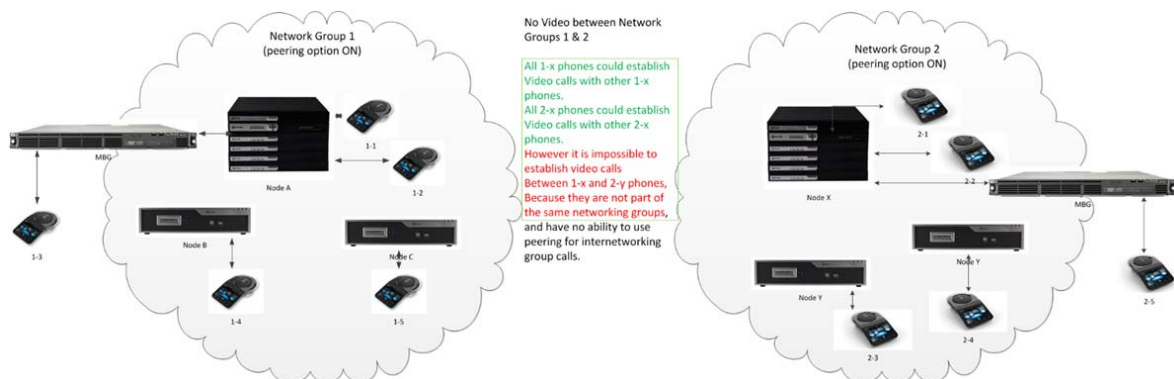
Configure Database Programming as follows to support SIP Peer-to-Peer video:

1. Create the SIP phones.

2. Create a new Network Group.

3. In the Network Group, enable the **Use Peer-to-Peer Media** flag.

4. Move the SIP phones into the new Network Group.

See Database Programming Help for additional details.

Note the following limitations:

- SIP Endpoints must have the same video codecs for the video call.

- MiVoice Office 250 does not support video conferencing. Video conferencing is available if the SIP Endpoints can create it. See "Video Confernce Calls" on page 690.

- Note that the two devices must be in the same Network Group with the **Peer to Peer Media** flag enabled in order for a call to negotiate a SIP P2P video stream.

VIDEO CONFERNCE CALLS

The MiVoice Office 250 handles audio and video conferences differently. Audio conferences are hosted on the MiVoice Office 250 controller itself, and each conference participant is able to add another participant to this conference. Video conferences however, are hosted on the SIP client devices themselves. Effectively each participant would be able to set up a video conference on their SIP devices. For example, if another participant in the video conference uses their SIP device to create a new video conference, it is established and hosted on this participant's SIP device. From the original conference view it appears as if a conference-in-a-conference is taking place, and would be confusing for all participants. To overcome this limitation, ensure all participants are invited to a video conference from one SIP device only.

SERVICE PROVIDER SIP TRUNKS AND SIP TRUNK GROUPS

This section contains the following information:

- **Feature Description** below

- Information Required by SIP Trunk Service Providers on **page 695**

- Creating SIP Trunk Groups on **page 695**

- Configuring SIP Trunk Groups on **page 697**

- Troubleshooting SIP Trunk Service Providers on **page 699**

FEATURE DESCRIPTION

This section contains the following SIP peer trunk group-specific information:

- **SIP Peer Trunk Features** below

- **SIP Peer Trunk Group Features** below

- Service Provider SIP Trunks on **page 692**

- SIP Peer Trunk Group Registration on **page 692**

- SIP Peer Trunk Group Authentication on **page 694**

- SIP Peer Trunk Group Access Codes on **page 694**

SIP PEER TRUNK FEATURES

A SIP trunk is a virtual trunk that you must associate with a SIP peer trunk group. Unlike a SIP peer voice mail, only one call to each SIP trunk is allowed. You must define a SIP trunk for each call. A SIP trunk has a dialable extension, used for the Open Architecture Application (OAI). By having an extension, OAI can differentiate between each end of the call when a call comes in on a SIP peer trunk group, and then it is forwarded or transferred back out the same SIP peer trunk group.

For incoming calls, the MiVoice Office 250 picks the SIP trunk with the lowest extension number that is available. For outgoing calls, you can dial the SIP peer trunk group extension either directly or indirectly through Automatic Route Selection (ARS), or you can dial the SIP trunk directly. If you dial the SIP peer trunk group extension, the MiVoice Office 250 selects the SIP trunk with the highest extension number that is available. If you dial the SIP trunk directly, the call could camp-on if the corresponding SIP trunk is busy. If the SIP trunk does not have a license, then the MiVoice Office 250 considers the SIP trunk an off-line device.

You can program a SIP trunk or SIP peer trunk group as a Direct Station Selection/Busy Lamp Field (DSS/BLF)/Programmable Key Modules (PKM) button. An individual SIP trunk provides status to a DSS/BLF (PKM) button based on the SIP peer trunk group instead of the individual SIP trunk.

SIP PEER TRUNK GROUP FEATURES

A SIP peer trunk group provides Central Office (CO) access using SIP. This functionality enhances the existing support for SIP trunks. A SIP peer trunk group's features are similar to a CO trunk group, which has a dialable extension in a CO trunk group range. You can put the SIP peer trunk group in an ARS facility group, and if dialed directly, it provides dial tone and prompts the caller for an outside number. The Phone and Voice Mail Administrator can assign the SIP trunk to a SIP peer trunk group through the administrator phone. The SIP trunk maintains the same OAI model as the existing CO Trunk Group and Trunk.

SERVICE PROVIDER SIP TRUNKS

Service provider SIP (SIP peer) trunks allow the MiVoice Office 250 to communicate with the CO using SIP trunks provided by a service provider without using a gateway. Service provider SIP trunks provide almost the same functionality as gateway SIP trunks, but they may require the MiVoice Office 250 to register and authenticate with the SIP trunk service provider network.

The MiVoice Office 250 supports various SIP Trunk service providers. For an updated list of supported SIP trunk service providers, refer to the Mitel SIP Center of Excellence on Mitel Online.

Configuration templates are included with the DB Programming installation, which are used to create and configure the SIP peer trunk group for the supported SIP trunk service providers (see **page 695** for programming). The existing SIP Peers folder contains SIP Voice Mails.

After you configure the SIP peer trunk group in DB Programming and registration is initialized with the SIP peer network, the SIP peer provisions the account information (for example, the user name and password) for the MiVoice Office 250 and provides that information to you. To authenticate the account, you must configure this information for the SIP peer in DB Programming (see **page 719**).

SIP PEER TRUNK GROUP REGISTRATION

The SIP peer (that is, the SIP trunk service provider) may require the MiVoice Office 250 to register with the SIP peer network using SIP Registrations. After you configure the SIP peer trunk group in DB Programming and registration is initialized with the SIP peer network, the SIP peer provisions the account information (for example, the user name and password) for the MiVoice Office 250 and provides that information to you. To authenticate the account, you must configure this information in the Authentication folder.

If the SIP peer does not require the MiVoice Office 250 to register itself with the SIP peer network, you need to create a static binding for the SIP peer in DB Programming. This static binding requires the public IP address or hostname, the User Datagram Protocol (UDP) Transport Protocol, and the listen port of the MiVoice Office 250. You must provide this information to the SIP peer along with contact information for the main representative at the customer's site (name, phone number, and e-mail address).

When you create a SIP peer trunk group in DB Programming which requires registration, the MiVoice Office 250 initiates SIP registration with the SIP peer network. If the SIP peer rejects the credentials from the MiVoice Office 250 registration request, then system alarm 144 (<#####> REG FAIL) generates, where <####> is the corresponding SIP peer trunk group extension. If the SIP peer accepts the credentials, a network binding is created on the SIP peer

network for the MiVoice Office 250. Also, the MiVoice Office 250 initiates a timer to refresh the registration, which is less than the registration expires value that the SIP peer network returns.

You need to create a static binding for the SIP peer in DB Programming if the SIP peer does not require the MiVoice Office 250 to register with the SIP peer network. You must provide the SIP peer with the following information about the MiVoice Office 250:

A static binding which requires the public IP address (System\IP Settings):

- Base Server

- Processor Module

- Processing Server (PS-1)

- Processor Expansion Card (PEC-1)

- SIP UDP Listening Port (5060)

- Main contact information (name, phone number, and e-mail address)

If the SIP peer network does not return a registration expiration value, then the system uses the configured refresh registration interval. The refresh registration is issued in about half of the time of this refresh interval. When the refresh timer expires, the MiVoice Office 250 initiates the SIP registration. This refreshes the network binding on the specified account (SIP peer trunk group).

When any of the following configuration parameters for the SIP peer trunk group is changed through DB Programming, the registration logic resets and results into new registration:

- IP Address

- Port Number

- Hostname

- Transport Type

- Domain Name

- NAT'ted IP Address

- Registrar Hostname

- Registrar IP Address

- Registrar Port Number

- Require Registration (If changed to disabled, then registration stops)

- Username

- Password

If the SIP registration request fails because of time-out, the registration request is retried using a back-off timer. If the SIP registration request still fails because of time-out, system alarm 144 (<#####> REG FAIL) is generated. To reinitiate the registration process, you can either change the operating state of the SIP peer in DB Programming from In-Service to Out-Of-Service Maintenance, and then back to In-Service, or change any of the parameters mentioned in the

list above. This resets the registration process, but the Phone Administrator must clear this alarm from the administrator phone.

See **page 718** for details about SIP peer registration programming.

SIP PEER TRUNK GROUP AUTHENTICATION

Some SIP peers require outbound calls to be authenticated. To obtain the information to authenticate the account (user name and password), some service providers have a Web site where you establish an account with them to view your user name, password, and other account information. You must program the user name and password into DB Programming. The service provider uses the user name and password to authenticate the account. You do not need to provide them with any information about the MiVoice Office 250.

You must provide the following information for the MiVoice Office 250 if the SIP peer does not require registration:

- A static binding which requires the public IP address (System\IP Settings in DB Programming):
  - Base Server
  - Processor Module
  - Processing Server (PS-1)
  - Processor Expansion Card (PEC-1)
- SIP UDP Listening Port (5060)
- Main contact information (name, phone number, and e-mail address)

For an outbound call, the MiVoice Office 250 sends a SIP invitation request to the SIP trunk service provider. If the SIP peer rejects the credentials from the MiVoice Office 250 invitation request, then system Alarm 146 (<#####> AUTH FAIL) is generated, where <####> is the corresponding SIP peer trunk group extension. If the SIP peer accepts the credentials, a network binding is created on the SIP peer network for the MiVoice Office 250, and then outbound calls are established. However, the Phone Administrator must clear this alarm from the administrator phone.

If the SIP invitation request fails for other reasons, the call failure count for the SIP peer trunk group could increase by one. If the failure count exceeds the configured maximum count, then the SIP peer changes to the Out-Of-Service operating state and system alarm 145 (<#####> OUT OF SVC) generates. This alarm clears automatically when the SIP peer changes to the In-Service operating state. You can change the operating state in DB Programming.

See **page 719** for details about SIP peer authentication programming.

SIP PEER TRUNK GROUP ACCESS CODES

Different access codes are supported by the SIP peer service providers. These providers prefer to have these numbers sent by the trunk to them without a "+" or "1" preceding the number. With 911 it is very important to use the correct Caller ID since this is how the provider knows what Public-Safety Answering Point (PSAP) to route the emergency call to. However, "+1911" and "+1411" works as well.

A SIP peer trunk group does **not** support the following CO Trunk Group functionality:

- Absorbed patterns

- One-way incoming call

- Echo trunk number

- Direct Inward System Access (DISA)

- Day/Night Answer Access

INFORMATION REQUIRED BY SIP TRUNK SERVICE PROVIDERS

To program SIP peer trunk groups, specific information is required by each SIP trunk service provider. See the Mitel SIP Center of Excellence *Compatibility and Third-Party Certification Reference Guide for Mitel Products*, available from Mitel Online.

For an updated list of supported SIP trunk service providers, refer to the Mitel SIP Center of Excellence on Mitel Online.

CREATING SIP TRUNK GROUPS

To support SIP trunks through a SIP trunk service provider, the SIP Trunk Groups folder has been added to the SIP Peers folder in DB Programming.

Each template contains predefined SIP peer trunk group settings specific to that SIP trunk service provider. Use the provided templates to create a SIP peer trunk group. You can also create and configure a SIP trunk group without using a template and then export those settings to use as a template to create additional SIP peer trunk groups. To make programming easier you can import and export these templates for use on other nodes.

Each SIP trunk group must have unique IP addresses or Fully Qualified Domain Names (FQDN). If you attempt to create a SIP trunk group with identical IP/FQDN values, the following error message appears.
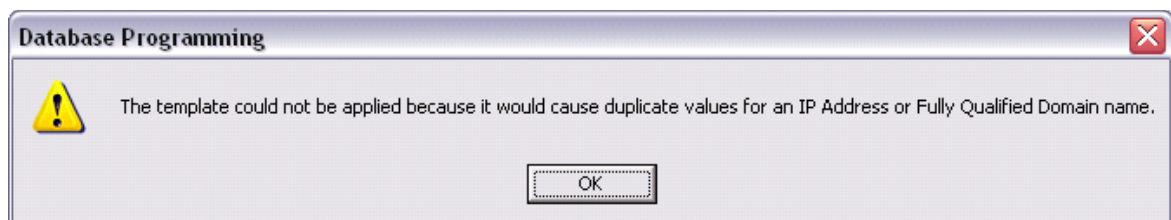
To set unique IP/FQDN values for each SIP trunk group, program the following fields in the Configuration folder:

- IP Address

- Fully Qualified Domain Name

- Alternate IP/FQDN List (for Bandwidth.com and NetSolutions only)

To create a SIP trunk group:

1. Select System – Devices and Feature Codes – SIP Peers – **SIP Trunk Groups**.

2. Right-click the right pane, and then click **Create SIP Trunk Group**. The Create SIP Trunk Group Extension dialog box appears.

3. Select the extension number you want to use for the item in the Starting Extension field. The range is 92001–92999; the default baseline extension for a SIP trunk group is based on the CO Trunk Group baseline of 92001.

4. Indicate the number of extensions you want to create in the **Number of Extensions** field. If the system is set to have more than one extension, the new SIP trunk groups are assigned sequentially to the next available numbers.

    Click **OK**. When you create a SIP trunk group, the Call Configurations subfolder under System\IP-Related Information creates a new call configuration for the SIP trunk group by default (see "IP Call Configurations" on **page 629** for details).

5. Assign a username and its description to the SIP trunk group. The description that appears in all trunk group lists in the database can be up to 20 characters long. The username that appears on display phones can have up to 10 characters.

To create a SIP trunk group using a template:

1. Select System – Devices and Feature Codes – SIP Peers – **SIP Trunk Groups**.

2. Right-click the right pane, and then click **Create SIP Trunk Group from Template**. The Select the Template to use dialog box appears. The dialog box opens to the Templates folder. The Templates folder is located in C:\Program Files\Mitel\5000\Templates (for v3.1 or later).

3. Select the template to use, and then click **Open**. The Create SIP Trunk Group Extension dialog box appears.

4. Select the extension number you want to use for the item in the Starting Extension field. The range is 92001–92999; the default baseline extension for a SIP trunk group is based on the CO Trunk Group baseline of 92001.

5. Indicate the number of extensions you want to create in the **Number of Extensions** field. If the system is set to have more than one extension, the new SIP trunk groups are assigned sequentially to the next available numbers.

Click **OK**. When you create a SIP trunk group, the Call Configurations subfolder under System\IP-Related Information creates a new call configuration for the SIP trunk group by default (see "IP Call Configurations" on **page 629** for details).

6. Assign a username and its description to the SIP trunk group. The description that appears in all trunk group lists in the database can be up to 20 characters long. The username that appears on display phones can have up to 10 characters.

7. After you create a SIP trunk group, right-click the trunk group to do any of the following as needed:

   • **Explore**: Configures the selected SIP trunk group (see **page 697**).

   • **Copy**: Copies the SIP trunk group configuration.

   • **Apply Template**: Applies a preconfigured template to a SIP trunk group.

   • **Export Settings as Template**: Exports SIP trunk group configuration settings to a template. When you select this menu option, a file save dialog box appears. Type a filename for the template, and then click Save. (If you select the original `.stg` file, you will overwrite the default settings. Mitel recommends that you rename the file.)

   • **Delete**: Removes all of the associated SIP trunks as well as the SIP trunk group. Deleting a SIP trunk group drops all calls on the deleted SIP trunk group.

CONFIGURING SIP TRUNK GROUPS

This section contains the following information:

• **Configuring the SIP Peer Trunk Group Programming Options** below

• **Programming SIP Peer Trunk List** below

• Programming SIP Peer Trunk Group Configuration on **page 698**

• Configuring SIP Peer Trunk Groups for Emergency Calls on **page 699**

When you use a template to create a SIP peer trunk group, all of the necessary fields specific to the SIP trunk service provider are automatically configured. The configuration is preprogrammed for you and you do not need to change any settings (except you need to add a user name and password for VoiceFlex after the template is applied). When you create a SIP peer trunk group without using a template, you must obtain the necessary information from the SIP trunk service provider, and then configure this information in DB Programming. The following section details the fields that must be configured and the specific settings for SIP trunk service providers. These settings might change if the SIP trunk service provider change their settings, or provide you with configurations other than what are described in this section.

To configure the SIP trunk group:

1. Select System – Devices and Feature Codes – SIP Peers – SIP Trunk Groups – <*SIP trunk group*> – **Configuration**.

2. Set the settings specific to a SIP trunk service provider.

3. Select System – Devices and Feature Codes – SIP Peers – SIP Trunk Groups – <*SIP trunk group*> – **Trunk Group Configuration**.

4. Set general settings for a SIP peer trunk group (see below).

CONFIGURING THE SIP PEER TRUNK GROUP PROGRAMMING OPTIONS

Program the Configuration folder as described on **page 717**.

PROGRAMMING SIP PEER TRUNK LIST

Individual SIP peer trunks must reside in a SIP peer trunk group. A SIP peer trunk group provides virtually the same functionality as a CO Trunk Group.

To program SIP peer trunk list:

1. Select System – Devices and Feature Codes – SIP Peers – SIP Trunk Groups – *<SIP trunk group>* – **Trunk Group Configuration**.

2. Create the SIP peer trunks as follows:

   a. Double-click **Trunks**.

   b. Right-click the right pane, and the select **Create SIP Peer Trunk**. The Create SIP Peer Trunk Extension dialog box appears.

   c. Select the extension number you want to use for the item in the Starting Extension field. The range is 94001–94999; the default baseline start extension for a SIP peer trunk is based on the Trunk baseline of 94001.

   d. Indicate the number of extensions you want to create in the Number of Extensions field. If the system is set to have more than one extension, the new trunks are assigned sequentially to the next available numbers.

   e. Click **OK**. For each SIP peer trunk that you create, the **Maximum Number of Calls** (in the Configuration folder for the SIP peer trunk group) increments by one. The number of SIP peer trunks is restricted by the number of available SIP Trunks licenses.

   f. After you create the SIP peer trunks, right-click the trunk to do any of the following as needed:

      • **Delete**: Removes the associated SIP trunk. For each SIP peer trunk that you delete, the Maximum Number of Calls (in the Configuration folder for the SIP peer trunk group) decrements by one.

      • **Batch Extension Change**: Changes multiple extension numbers at one time.

PROGRAMMING SIP PEER TRUNK GROUP CONFIGURATION

See "Programming Trunk Group Options" on **page 506** or refer to the *MiVoice Office 250 DB Programming Help* for SIP trunk group programming (System\Devices and Feature Codes\SIP Peers\SIP Trunk Groups*<trunk group>*\Trunk Group Configuration) and details about these fields.

The only field you must configure different from the default setting is the **Force Trunk Group Calling Party Name and Number**. By default this is set to No.

NOTE	For detailed information about Caller ID forwarding and priorities, see "Caller ID Forwarding" on **page 534**.

See **page 699** for details on troubleshooting service provider SIP trunks.

CONFIGURING SIP PEER TRUNK GROUPS FOR EMERGENCY CALLS

Emergency calls (for example, 911, 999, 000, and 112) will fail because the Call Failure
Threshold, in System\Devices and Feature Codes\SIP Peers\SIP Trunk Groups\<*SIP trunk
group*>\Configuration in OLM, must be met before the system sources out other resources.
See "Emergency Extensions for IP Devices" on **page 642** for complete details about emergency
calling configuration. To avoid failed emergency calls, follow one of the recommended
configurations:

Configuration 1

Decrease the Keep-Alive Ping Interval for the SIP peer trunk group. This takes the SIP peer
trunk group out of service as soon as the peer is unreachable and emergency calls will fallback
to the next facility group.

NOTE	Emergency calls made between two consecutive keep-alives will fail if the SIP peer becomes unreachable during that time.

Configuration 2

Emergency calls are always routed through the first route group:

1. Create a route group for 911, 999, 000, or 112 (dial pattern as 911, 999, 000, or 112) and
   place this route group first in the list of route groups.

2. Add a facility group with a PRI or Loop start (which immediately goes out of service unlike
   a SIP peer trunk group) and place this facility group at the top of the list.

   The second facility group can be the SIP peer trunk group.

   The next route group can be used for all other calls.

> ### NOTICE
> Emergency Call phone numbers include:
> - 911, the default for Mitel systems located in the United States (USA).
> - 999, the default for Mitel systems located in the European market and used primarily in the United Kingdom (UK).
> - If applicable, 112, an emergency number used widely in Europe outside of the UK, and in Australia.
> - Any emergency number, such as for a police or fire station, that is appropriate for the location of the main system and/or remote phones.

TROUBLESHOOTING SIP TRUNK SERVICE PROVIDERS

> ### NOTICE
> **Network Troubleshooting**: Maintenance and troubleshooting of your LAN/WAN network is the responsibility of your network provider. Mitel Technical Support can help you isolate minor network problems; Technical Support will escalate complex network problems to Professional Services, a billable service.

This section provides examples of a working scenario SIP call flow and Message Print output along with troubleshooting information for the following:

- **Registration** below

- Basic Incoming Calls on **page 701**

- Unauthenticated Basic Outgoing Calls on **page 703**

- Authenticated Basic Outgoing Calls on **page 703**

REGISTRATION

Some SIP trunk service providers, such as VoiceFlex, require the MiVoice Office 250 to register with the provider using SIP. This section details information about registration. See the following:

- The figure below is a SIP call flow example for a successful registration.

```
| Mitel 5000          | VoiceFlex           |
| 172.16.15.87        | 146.101.248.200     |
|                     |                     |
|        Request: REGISTER                   |SIP: Request: REGISTER sip:146.101.248.200:5060
|(5060)  ------------------->  (5060)  |
|        Status: 100 Trying                  |SIP: Status: 100 Trying     (1 bindings)
|(5060)  <------------------  (5060)  |
|        Status: 401 Unauth                  |SIP: Status: 401 Unauthorized    (0 bindings)
|(5060)  <------------------  (5060)  |
|        Request: REGISTER                   |SIP: Request: REGISTER sip:146.101.248.200:5060
|(5060)  ------------------->  (5060)  |
|        Status: 100 Trying                  |SIP: Status: 100 Trying     (1 bindings)
|(5060)  <------------------  (5060)  |
|        Status: 200 OK                      |SIP: Status: 200 OK     (0 bindings)
|(5060)  <------------------  (5060)  |
```

- The figure below is an example of Message Print for a successful registration with SIP header information enabled.

```
)7:734- 09:18 10-17 *** SIP TX ['''-'92003']: REGISTER sip:146.101.248.200:5060 SIP/2.0
)7:735- 09:18 10-17 *** SIP RX ['''-'92003']: SIP/2.0 100 Trying
)7:736- 09:18 10-17 *** SIP RX ['''-'92003']: SIP/2.0 401 Unauthorized
)7:737- 09:18 10-17 *** SIP TX ['''-'92003']: REGISTER sip:146.101.248.200:5060 SIP/2.0
)7:738- 09:18 10-17 *** SIP RX ['''-'92003']: SIP/2.0 100 Trying
)7:739- 09:18 10-17 *** SIP RX ['''-'92003']: SIP/2.0 200 OK
```

- Table 116 on **page 701** lists troubleshooting information for registering with the SIP trunk service provider.

Table 116:   Service Provider SIP Trunks Registration Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Registration with the SIP trunk service provider failed and alarm 144 (REG FAIL) has generated.	The MiVoice Office 250 could not contact the service provider.	• Verify that the IP addresses and ports are configured correctly in DB Programming. (See Example 1 below.) • Verify network connectivity.
	The authentication parameters are invalid.	Verify that the user name and password provided by SIP trunk service provider are programmed correctly in DB Programming. (See Example 2 below.)

Example 1. The following is a snippet from Message Print when the SIP trunk service provider (VoiceFlex) IP address is programmed incorrectly:

```
-07:004- 10:31 10-17 *** SIP RX ['''-'''']: INVITE sip:+441376653001@172.16.15.87:5060 SIP/2.0
-07:005- 10:31 10-17 *** SIP TX ['''-'''']: SIP/2.0 403 Forbidden
-07:006- 10:31 10-17 *** SIP RX ['''-'''']: ACK sip:+441376653001@172.16.15.87:5060 SIP/2.0
```

Example 2. The following is a snippet from Message Print when the username or password is programmed incorrectly:

```
-07:842- 09:50 10-17 *** SIP TX [''-'92003']: REGISTER sip:146.101.248.200:5060 SIP/2.0
-07:844- 09:50 10-17 *** SIP RX [''-'92003']: SIP/2.0 100 Trying
-07:845- 09:50 10-17 *** SIP RX [''-'92003']: SIP/2.0 401 Unauthorized
-07:846- 09:50 10-17 *** SIP TX [''-'92003']: REGISTER sip:146.101.248.200:5060 SIP/2.0
-07:848- 09:50 10-17 *** SIP RX [''-'92003']: SIP/2.0 100 Trying
-07:849- 09:50 10-17 *** SIP RX [''-'92003']: SIP/2.0 403 Forbidden (Bad auth)
```

BASIC INCOMING CALLS

This section details information for basic incoming calls. See the following:

• The figure below is a SIP call flow example for basic incoming calls.

```
| Mitel 5000        | VoiceFlex        |
| 172.16.15.87      | 146.101.248.200  |
|                   |                  |
|        Request: INVITE sip           |SIP/SDP: Request: INVITE sip:+441376653001@172.16.15.87:5060
|(5060)    <------------------  (5060) |
|        Status: 100 Trying            |SIP: Status: 100 Trying
|(5060)    ------------------>  (5060) |
|        Status: 180 Ringing           |SIP: Status: 180 Ringing
|(5060)    ------------------>  (5060) |
|        Status: 200 OK, wit           |SIP/SDP: Status: 200 OK, with session description
|(5060)    ------------------>  (5060) |
|        Request: ACK sip:17           |SIP: Request: ACK sip:172.16.15.87:5060
|(5060)    <------------------  (5060) |
|        Request: BYE sip:17           |SIP: Request: BYE sip:172.16.15.87:5060
|(5060)    <------------------  (5060) |
|        Status: 200 OK                |SIP: Status: 200 OK
|(5060)    ------------------->  (5060)|
```

- The figure below is an example of Message Print for basic incoming calls with SIP header information enabled.

```
-07:963- 10:18 10-17 *** SIP RX ['' -'']: INVITE sip:+441376653001@172.16.15.87:5060 SIP/2.0
-07:964- 10:18 10-17 *** SIP TX ['94008'-'@03*P']: SIP/2.0 100 Trying
-07:965- 10:18 10-17 *** SIP TX ['94008'-'@03*P']: SIP/2.0 180 Ringing
-07:966- 10:18 10-17 *** SIP TX ['94008'-'@03*P']: SIP/2.0 200 OK
-07:967- 10:18 10-17 *** SIP RX ['94008'-'@03*P']: ACK sip:172.16.15.87:5060 SIP/2.0
-07:968- 10:18 10-17 *** SIP RX ['94008'-'@03*P']: BYE sip:172.16.15.87:5060 SIP/2.0
-07:969- 10:18 10-17 *** SIP TX ['94008'-'@03*P']: SIP/2.0 200 OK
```

- **Table 117** below lists troubleshooting information for basic incoming calls.

Table 117:   Service Provider SIP Trunks Basic Incoming Call Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
The SIP trunk group is not receiving calls.	The MiVoice Office 250 is not registered with the SIP trunk service provider who requires registration.	Contact the SIP trunk service provider to get registered.
	The SIP trunk service provider is sending SIP messages from an IP address that the MiVoice Office 250 does not know about. The MiVoice Office 250 sends a 403 Forbidden SIP message.	Verify that the IP address provided by the SIP trunk service provider is programmed in the "Alternate IP Address" for the SIP trunk group. (See Example 1 below).
	There are problems with the network connection.	Verify network connectivity.

Example 1. The following is a snippet from Message Print when the MiVoice Office 250 receives a SIP message from an unknown IP address:

```
-07:004- 10:31 10-17 *** SIP RX ['' -'']: INVITE sip:+441376653001@172.16.15.87:5060 SIP/2.0
-07:005- 10:31 10-17 *** SIP TX ['' -'']: SIP/2.0 403 Forbidden
-07:006- 10:31 10-17 *** SIP RX ['' -'']: ACK sip:+441376653001@172.16.15.87:5060 SIP/2.0
```

UNAUTHENTICATED BASIC OUTGOING CALLS

Some SIP trunk service providers, such as Bandwidth.com, do not authenticate outgoing calls. The service providers look at the IP address and the call succeeds if the IP address of the MiVoice Office 250 they have in their records matches the source IP of the SIP message. This section details information for unauthenticated basic outgoing calls. See the following:

- The figure below is a SIP call flow example for unauthenticated basic outgoing calls.

```
| Mitel 5000        | Bandwidth.com       |
| 65.166.206.212    | 216.82.224.202      |
|                   |                     |
|        Request: INVITE sip              |SIP/SDP: Request: INVITE sip:+14809619000@216.82.224.202:5060
|(5060)  ------------------>  (5060)      |
|        Status: 100 Giving                |SIP: Status: 100 Giving a try
|(5060)  <------------------   (5060)     |
|        Status: 200 OK, wit               |SIP/SDP: Status: 200 OK, with session description
|(5060)  <------------------   (5060)     |
|        Request: ACK sip:19               |SIP: Request: ACK sip:199.173.94.144:5060
|(5060)  ------------------>  (5060)      |
|        Request: BYE sip:19               |SIP: Request: BYE sip:199.173.94.144:5060
|(5060)  ------------------>  (5060)      |
|        Status: 200 OK                    |SIP: Status: 200 OK
|(5060)  <------------------   (5060)     |
```

- **Table 118** below lists troubleshooting information for unauthenticated basic outgoing calls.

Table 118:   Service Provider SIP Trunks Unauthenticated Outgoing Calls Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Outgoing calls from the SIP trunk group are failing.	There are problems with the network connection.	Verify network connectivity.
	The SIP trunk service provider is unavailable.	If there are no network connectivity issues, contact the SIP trunk service provider to make sure their services are operational.

AUTHENTICATED BASIC OUTGOING CALLS

Some providers, such as VoiceFlex, authenticate all outgoing calls. This section details information for authenticated basic outgoing calls. See the following:

- The figure below is a SIP call flow example for authenticated basic outgoing calls.

```
-07:754- 09:22 10-17 *** SIP TX ['94009'-'@0391']: INVITE sip:+14809619000@146.101.248.200:5060
SIP/2.0
-07:755- 09:22 10-17 *** SIP RX ['94009'-'@0391']: SIP/2.0 407 Proxy Authentication Required
-07:756- 09:22 10-17 *** SIP TX ['94009'-'@0391']: ACK sip:+14809619000@146.101.248.200:5060
SIP/2.0
-07:757- 09:22 10-17 *** SIP TX ['94009'-'@0391']: INVITE sip:+14809619000@146.101.248.200:5060
SIP/2.0
-07:758- 09:22 10-17 *** SIP RX ['94009'-'@0391']: SIP/2.0 100 Trying
-07:759- 09:22 10-17 *** SIP RX ['94009'-'@0391']: SIP/2.0 183 Session Progress
-07:760- 09:22 10-17 *** SIP RX ['94009'-'@0391']: SIP/2.0 200 OK
-07:761- 09:22 10-17 *** SIP TX ['94009'-'@0391']: ACK sip:+14809619000@146.101.248.200 SIP/2.0
-07:764- 09:22 10-17 *** SIP TX ['94009'-'@0391']: BYE sip:+14809619000@146.101.248.200 SIP/2.0
-07:765- 09:22 10-17 *** SIP RX ['94009'-'@0391']: SIP/2.0 200 OK
```

- The figure below is an example of Message Print for authenticated basic outgoing calls with SIP header information enabled.

```
| Mitel 5000       | Bandwidth.com    |
|  65.166.206.212  |  216.82.224.202  |
|                  |                  |
|       Request: INVITE sip           |SIP/SDP: Request: INVITE sip:+14809619000@216.82.224.202:5060
| (5060)  ------------------>  (5060) |
|       Status: 100 Giving            |SIP: Status: 100 Giving a try
| (5060)  <------------------  (5060) |
|       Status: 200 OK, wit           |SIP/SDP: Status: 200 OK, with session description
| (5060)  <------------------  (5060) |
|       Request: ACK sip:19           |SIP: Request: ACK sip:199.173.94.144:5060
| (5060)  ------------------>  (5060) |
|       Request: BYE sip:19           |SIP: Request: BYE sip:199.173.94.144:5060
| (5060)  ------------------>  (5060) |
|       Status: 200 OK                |SIP: Status: 200 OK
| (5060)  <------------------  (5060) |
```

- **Table 119** below lists troubleshooting information for authenticated basic outgoing calls.

Table 119: Service Provider SIP Trunks Authenticated Outgoing Calls Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Authentication with the SIP trunk service provider failed and alarm 146 (AUTH FAIL) has generated.	The authentication parameters are invalid.	Verify that the user name and password provided by the SIP trunk service provider are programmed correctly in DB Programming. (See Example 1 below).

Example 1. The following is a snippet from Message Print when the username or password is programmed incorrectly:

```
-07:913- 10:11 10-17 *** SIP TX ['94009'-'@03*4']: INVITE sip:+14809619000@146.101.248.200:5060
SIP/2.0
-07:914- 10:11 10-17 *** SIP RX ['94009'-'@03*4']: SIP/2.0 407 Proxy Authentication Required
-07:915- 10:11 10-17 *** SIP TX ['94009'-'@03*4']: ACK sip:+14809619000@146.101.248.200:5060
SIP/2.0
-07:916- 10:11 10-17 *** SIP TX ['94009'-'@03*4']: INVITE sip:+14809619000@146.101.248.200:5060
SIP/2.0
-07:917- 10:11 10-17 *** SIP RX ['94009'-'@03*4']: SIP/2.0 403 Forbidden
-07:918- 10:11 10-17 *** SIP TX ['94009'-'@03*4']: ACK sip:+14809619000@146.101.248.200:5060
SIP/2.0
```

Additional troubleshooting information currently exists for the SIP peer voice mail, which was added in the v3.0 or later software. This information applies to the SIP peer trunk group (see **page 691**):

- Operating state

- Status

- SIP view diagnostics feature code 9987 [9187]

- System Monitor dumps

- Log files

MBG SIP SERVICES – TRUNKS AND DEVICES

The MBG (Mitel Border Gateway) is a platform for the secure deployment of multiple network connectivity services in a number of network edge scenarios:

- Teleworker Service: A secure plug & play solution for remote workers. SIP device support was added in this service pack release, in addition to the MiNet device support available in release 6.0. (Depending on whether an MBG is connected to the MiVoice Office 250 network, and remote 53xx series phones are outside the network, the appropriate NAT Address Type settings need to be programmed.)

- SIP Trunk Proxy Service: Serves a SIP-aware firewall at the edge of the company network and eliminates the need for 3rd party firewalls, and simplifies configuration and deployment.

- Application Web Proxy Service: Enables trusted connectivity between the company LAN and the Public Internet to provide secure access for unified communication, voice mail and web portal services via the 5000 system and Mitel Applications Suite (MAS).

Figure 7: Example MBG Network Diagram
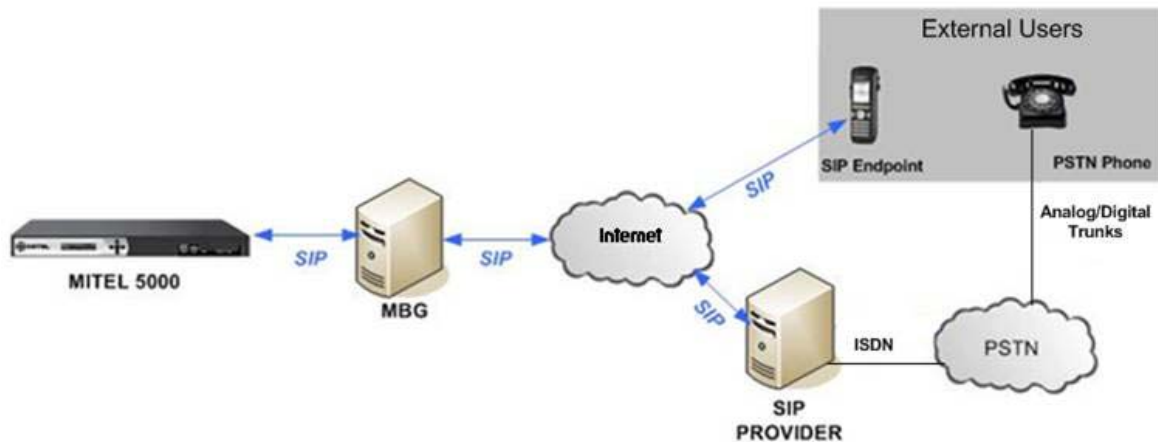
> **NOTE** Refer to the SIP CoE document for information on supported SIP devices.

By connecting to the MiVoice Office 250 through the MBG, external users can access many MiVoice Office 250 features available to internal users.

SIP TRUNK CALL FEATURES

SIP Trunk Call Features supported with MBG include the following:

- Conference
- Forwarding Always/Busy/No Answer
- Transfer Supervised and Unsupervised
- Transfer Recall
- Transfer to Forward SIP

- Hold

- Music-On-Hold

- Dynamic Extension Express and Mid-Call Features

- Mute

- Redial

- SIP Calling Number and Name Delivery

- Do-Not-Disturb

- Incoming/Outgoing Call with G.711 and G.729

- ACD calls

SYSTEM REQUIREMENTS

- To use this functionality, the system must meet these requirements:

- MiVoice Office 250 6.0 SP1 and higher

- MBG GA 8.0.17.0 and higher

- MAS SP1 application (5.0.116.0) that includes:

  - UCA SP1 - 6.0.120.0 and higher
  - MCA SP1 - 5.0.1.17 and higher
  - MiCollab Unified Messaging 16.1.0.13 and higher

5000 LICENSING AND MBG

There are no new MBG licenses or new 5000 licenses for MBG support. The appropriate 5000 and MBG licenses are required for the devices and services to function.

CONFIGURING MIVOICE OFFICE 250/MBG SIP TRUNKING

| **NOTE** | For configuration information specific to your SIP Service Provider, refer to the MSA SIP Center of Excellence information on Mitel Online (http://ebiz.mitel.com/images/sipcoe/SIP_CoE.pdf). |

To use SIP Trunking there must be proper configuration of SIP Trunks between MiVoice Office 250 system and MBG and between MBG and the PSTN. This includes:

- Configuring the 5000 Nodes on the MBG

- Configuring SIP Trunks in MiVoice Office 250 DP Programming

CONFIGURING THE 5000 NODES ON THE MBG

There must be a valid license on the MBG. Refer to the *MBG Installation & Maintenance Guide* for more detailed information on programming the 5000 nodes on the MBG.

CONFIGURING SIP TRUNKS IN MIVOICE OFFICE 250 DP PROGRAMMING

1. In Database Programming, go to *System/Devices and Feature Codes/SIP Peers/SIP Trunk Groups.*

2. Right click and select **Create SIP Trunk Group**. The *Create SIP Trunk Group Extension* dialog box appears.

3. Enter a trunk number for the **Starting Extension** number, select the **Number of Extensions** you want to create.
    Click **OK**.

4. Open the new SIP Trunk Group, and then select **Configuration**.

5. Click **IP Address**, enter the address of the SIP service provider in the **Edit IP Address** dialog box, and click OK.

6. Click **Fully Qualified Domain Name** and enter the FQDN of the SIP provider.

7. Open the **Route Sets** folder.

8. Add a Route Set:

    a. Right click in the right-side pane.

    b. Select **Add to Route Sets List**.

    c. When the Route Set appears, enter the Hostname or IP Address of the MBG (internal IP address or FQDN) in the **Hostname** field.

9. Open the **Trunk Group Configuration** folder for the SIP Trunk Group and enter the following:

    - **Calling Party Name**: Enter the name of the SIP service provider.

    - **Calling Party Number**: Enter the calling number of the SIP Service Provider.

10. Open the **Trunks** folder.

11. Right click in the right-side pane and select **Create SIP Peer Trunk**.
    The *Create SIP Peer Trunk Extension* dialog box appears.

12. Enter the **Starting Extension** number and the **Number of Extensions** you want to create.
    Click OK. (Create the number of SIP trunks according to the number of licenses.)

CONFIGURING MIVOICE OFFICE 250/UCA WITH MBG

UCA supports the following clients when used with MBG:

• Desktop phone (MiNet version) and Windows Desktop Softphone (MiNet version)

• SIP Softphone

• Mobile phone using UCA SIP Softphone (iOS and Android)

NOTE	While a Mobile device that is connected to UCA is within the office area connection, they have direct access to the MiVoice Office 250 features. However, as soon as they leave the office area, the roaming function is used and the mobile device is connected to the MBG. From that moment they become an external user, which has access to MiVoice Office 250 features through MBG.

The following MiVoice Office 250 system software licenses must be enabled to provide full integration with UC Advanced:

- System OAI 3rd Party Call Control

- System OAI Events

- Dynamic Extension Express (DEE)

To allow UCA to work with MBG, you must configure:

- Mitel 500 DB Programming

- UCA

MIVOICE OFFICE 250 CONFIGURATION

The UC Server retrieves information programmed in the Users and/or Stations, Endpoints or Phones folders to create UC Advanced accounts.

Go to *System/Devices and Feature Codes/Phones* and create the following:

- IP/Digital/Single Line/SIP phone for main extension

- UC Advanced Softphone for Desktop Client

- SIP phone for Mobile Client

*If the **Phones** information is configured for the MiVoice Office 250, but the **Users** information is not*

The UC Advanced PBX node synchronizer retrieves the following information from the **Phones** folder in DB Programming to create UC Advanced accounts:

- **Extension**: The value configured in the Extension field becomes the phone extension for the UC Advanced account after synchronization.

- **Description**: The value configured in the Description field becomes the account name for the UC Advanced account after synchronization. Note the following for the Description field:

  - Configure the user's name in Last name, First Name format for the Description field. This is the format used by the PBX node synchronizer for the account name in UC Advanced.

  - The Tilde character (~) before the Description and a blank Description field excludes the account from the synchronization.

*If the **Users** information is configured for the MiVoice Office 250, but the **Phones** information is not*

The UC Advanced PBX node synchronizer retrieves the following information from the **Users** folder in DB Programming to create UC Advanced accounts:

- **First Name** and **Last Name**: These fields provide the value for the account name on the UC Server. Note the following:

  - To exclude a UC Advanced account from being created during synchronization, insert the Tilde character (~) before the First Name or Last Name in the Phones, Endpoints or Stations folder.

  - If the First Name and Last Name fields are blank in DB Programming, UC Advanced

will not create an account.

*If both **Phones** and **Users** are programmed in DB Programming*

If the **Phones Extension** fields and **Users Main Extension** field match, then only one account is created.

Configure the following for each user:

- **Main Extension**: The value configured for the Main Extension field becomes the desk phone for the UC Advanced account after synchronization. (UC Advanced Softphones are allowed as the Main Extension.)

- **Associated Destinations**: Configure the values for the Associated Destinations sub folder:
    - **Softphone**: UC Advanced Softphone.
    - **Softphone 2**: SIP phone that has been created for Mobile Client.

For SIP phone compatibility with MBG

In *System/Devices and Feature Codes/SIP Peers/SIP Phone Groups* set the following:

- **Use Registered Username**: Select **Yes** for **Use Registered Username** value for the SIP phone profile group.

To support UC Advanced

Go to *System/Sockets*.

Select **Yes** in **System OAI Level 2**.

Passcode/Passwords

The SIP phones on the 5000 have two areas for setting the SIP phone password:

- Phone/Station/Endpoint password
- Authentication is under the SIP profile Group

Either password setting on the 5000 can be used with the MBG.

UCA CONFIGURATION

Refer to the *Unified Communicator Advanced Administration Guide* for details about configuring the UCA server with the MiVoice Office 250.

MBG CONFIGURATION

Ensure there are valid Teleworker (TW), SIP trunk and compression licenses on the MBG. The MBG allocates TW licenses for MiNet and SIP dynamically when the devices are online.

When a device is off-line the license is available for use where the licenses are not dedicated. If all MiNet and SIP devices users are active at the same time on a regular basis then the system will need to be configured with same amount of TW licenses as there are devices.

Refer to the *Mitel Border Gateway Installation and Maintenance Guide* for details.

CONFIGURING UCA MOBILE

After installing UCA Mobile, enter the UC Server hostname the first time you run it. This value can be modified later. If you have installed an older version, you will be asked to upgrade after your first login. Your login and password have already been configured.

In *Settings/Softphone Settings/Teleworker* settings make sure **Server**, **Username** and **Password** are configured (you have configured them at MBG) and the **Use Teleworker** checkbox is marked.

For correct behavior of SIP softphone with MBG and 5000, **Settings/Softphone Settings/ Advanced settings/Protocol** value should be **UDP**.

SIP VOICE MAILS

SIP voice mails support SIP voice processing systems (for example, MiCollab Unified Messaging).

This section contains the following information:

- **Creating SIP Voice Mail Devices** below

- Adding SIP Voice Mail Devices to the SIP Voice Mail Call Configuration on **page 712**

- Configuring SIP Voice Mails on **page 713**

CREATING SIP VOICE MAIL DEVICES

You can create SIP voice mail devices for voice processing applications.

To create a SIP Voice Mail device:

1. Select System – Devices and Feature Codes – SIP Peers – **SIP Voice Mails**.

2. Right-click the right pane, and then click **Create SIP Voice Mail**. A message appears asking if the SIP voice mail is for MiCollab Unified Messaging.

3. Click **Yes**. The Create SIP Voice Mail Extension dialog box appears.

NOTE	By default, you can create one SIP Voice Mail device.

4. In the **Starting Extension** box, select the starting extension number. The range is P9001–P9999; the default baseline start extension is P9001.

5. In the **Number of Extensions** box, enter the number of extensions that you want to create. If the system is set to have more than one extension, the new devices are assigned sequentially to the next available numbers.

6. Click **OK**.

7. Type a username and description for the device.

8. Right-click on the device to do any of the following:

   - **Explore**: Configures the selected device (see **page 713**).

   - **Copy**: This functionality is not used when the system is set to have only one SIP Voice Mail.

   - **Synchronize Mailboxes**: Synchronizes the mailbox data across all MiVoice Office 250 networked nodes. This ensures the mailbox data per SIP Peer stays accurate for OAI applications. The synchronization is only necessary for multi-node configurations where OAI applications are used within the multi-node environment.

     To synchronize mailboxes:

     Right-click on the device extension, and then select **Synchronize Mailboxes**. The Synchronize SIP Mailboxes dialog box appears to indicate the status of the synchronization. The duration of this synchronization depends on the number of mailboxes that exist, so a larger number could take a minute or so.
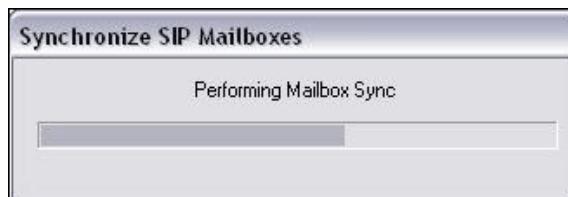
- **Delete**: Removes all of the associated SIP Voice Mail applications, mailboxes, and Group Lists as well as the SIP Voice Mail devices. Deleting Voice Mail devices drops all calls on the deleted devices.

ADDING SIP VOICE MAIL DEVICES TO THE SIP VOICE MAIL CALL CONFIGURATION

When you create a SIP Voice Mail (see **page 711**), the system creates a SIP Voice Mail Call Configuration. By default, the Call Configuration is named "MiCollab Unified Messaging" and the number is assigned sequentially to the next available number, as shown in the figure below.
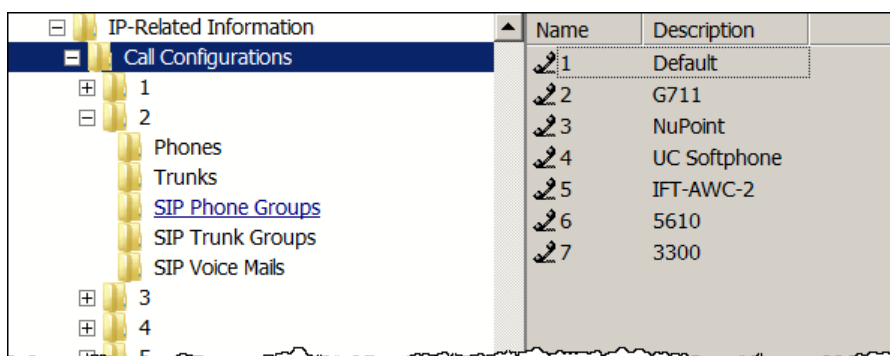
To add SIP Voice Mail devices to the SIP Voice Mail Call Configuration:

1. Click the SIP Voice Mail Call Configuration.

2. Double-click **SIP Voice Mails**. The right pane shows the SIP Voice Mail devices, if any, that are currently members of the Call Configuration.

3. Right-click anywhere in the right pane, and then select **Move To SIP Voice Mails List**.

4. Use the wizard to select the devices that use this Call Configuration. By default, you can add one device. The SIP Voice Mail device appears in the SIP Voice Mail folder as shown below. Double-click the SIP Voice Mail device to view and configure the SIP Voice Mail settings as described on **page 713**.
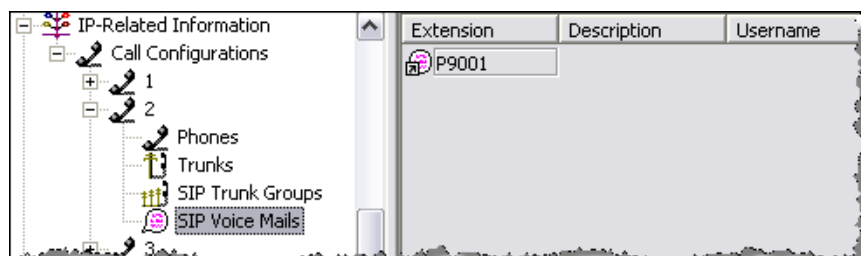
CONFIGURING SIP VOICE MAILS

The SIP Peer Voice Mail device options are located in System\Devices and Feature Codes\SIP Peers\SIP Voice Mails\<*SIP voice mail device*>.

To configure the SIP Voice Mail device:

1. Program the Configuration folder on **page 713**.

2. Create SIP Voice Mail applications on **page 713**.

3. Create group lists on **page 716**.

4. Create mailboxes on **page 716**.

PROGRAMMING THE SIP VOICE MAIL PROGRAMMING OPTION

Program the SIP Voice Mail Configuration folder as described on **page 717**.
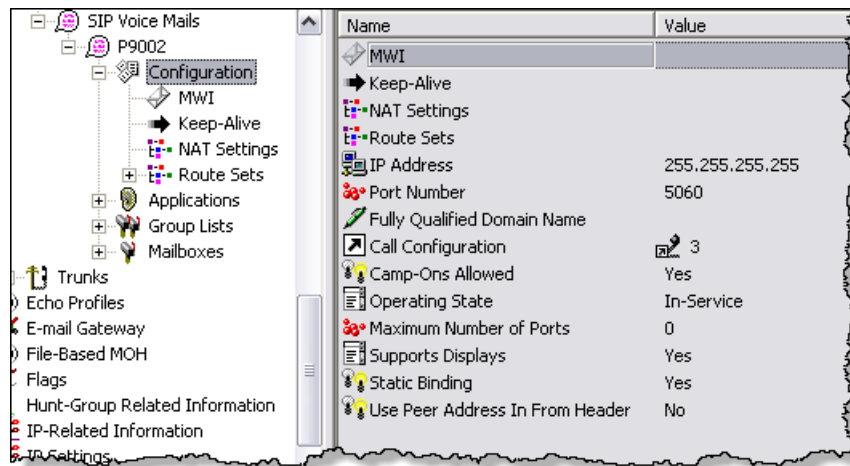
PROGRAMMING SIP VOICE MAIL APPLICATIONS

The Applications folder provides a list of the SIP Voice Mail applications associated with the SIP Peer. You can create the following SIP Voice Mail applications:

•   SIP Auto Attendant (AA)—see "Auto Attendant" on **page 874**

•   SIP Auto Attendant Recall (AAR)—see "Auto Attendant Recall" on **page 878**

•   SIP Call Routing Announcement (CRA)—see "Call Routing Announcements" on **page 879**

•   SIP Message Notification/Retrieval (MNR)*—see "Message Notification/Retrieval" on **page 890**

•   SIP Record-A-Call (RAC)—see "Record-A-Call" on **page 893**

•   SIP Scheduled Time-Based Application Router (STAR)—see "Scheduled Time-Based Application Routing (STAR)" on **page 911**

•   SIP Voice Mail (VM)—see "Voice Mail (Application)" on **page 915**

* Unlike MiVoice Office 250 Voice Mail, you can create multiple MNR applications.

To create a SIP Voice Mail application:

1. Right-click the right pane. A list of the SIP Voice Mail applications appears.

2. Select the application that you want to create. The **Create SIP *<application>* Extensions** dialog box appears.

   You must create a primary Voice Mail application. This must correspond to the Line Group created on the MiCollab Unified Messaging. For example, if Line Group 1 on the MiCollab Unified Messaging has a pilot number 2600, create an application with an extension 2600. Use this primary Voice Mail application as the pilot number for other applications, if needed (depending on the MiCollab Unified Messaging configuration).

3. Enter the extension number you want to use for the first item in the Starting Extension field.

4. Indicate the number of items you want to create in the Number of Extensions field. The new devices are assigned sequentially to the next available numbers.

5. Click **OK**.

6. Assign a Username and its description to the application.

7. After a Voice Mail application has been created, right-click the new device to do any of the following as needed:

   - **Explore**: Configures the selected application (see **page 715**).

   - **Copy and Paste**: Copies certain attributes from one application, and then pastes the attributes into one or more applications. The copy functionality only copies features that are shared among the selected application types.

   To copy an application:

   a. Right-click the application extension, and then select **Copy**.

   b. Right-click the application where you want the information pasted, and then select **Paste**.

   c. Select the attributes you want to copy from the Copy dialog box.

   d. Click **OK**.

      - **Delete**: Removes all of the selected applications.

      - **Batch Extension Change**: Changes several extensions at once.

   To change several extensions at once:

   a. Highlight the applications you want to change.

   b. Right-click, and then click **Batch Extension Change**.

   c. When the Create Extension screen appears, select the number you want to assign to the first highlighted feature code (the other highlighted codes will be numbered consecutively after this number).

   d. When you click **OK**, the extensions are automatically renumbered and re-sorted in the list.

To configure the application:

1. Click the application that you want to configure.

2. Configure the following fields:

   - SIP Voice Mail Pilot below

   - Prompt for Mailbox on Transfer below

   - Validate Mailbox on Transfer below

   - Class of Service on **page 400**

   - Attendants on **page 916**

   - Transfer Recall Destination on **page 918**

   - Music-On-Hold Parameters on **page 918**

     - Audio for Calls Camped onto this Device

     - Audio for Calls Holding for this Device

     - Audio for Calls Ringing this Device

   - Propagate Original Caller ID on Transfer on **page 919**

   - Calling Party Name and Number on **page 919**

   The following fields are SIP Voice Mail-specific options. The rest of the fields function the same as the existing fields for the MiVoice Office 250 Voice Mail applications:

   - **SIP Voice Mail Pilot**: Indicates the SIP Voice Mail pilot number. The pilot number must be set to a SIP VM application that is associated with a Line Group. This field allows MiCollab Unified Messaging to share the lines within a Line Group for calls to SIP Voice Mail applications. Typically, the majority of the MiCollab Unified Messaging lines reside in one MiCollab Unified Messaging Line Group. This MiCollab Unified Messaging Line Group has an associated extension on the MiVoice Office 250. This extension should be the "SIP Voice Mail Pilot" for all other SIP VM applications.

   To change the pilot, do one of the following methods:

   Method A

   a. Select the current value, and then type the new value in the text box.

   b. Press **ENTER**. A screen displays what is associated with the number you typed.

   c. Click **OK**. The new number appears in the field.

   Method B

   a. Right-click the current value field. Select **Change SIP Voice Mail Pilot**. A window appears prompting you to select the type to include. The SIP applications that appear as options for this field are on the parent/current SIP Peer only. You cannot select SIP applications that appear on another SIP Peer.

   b. Select the device(s), and then click **Next**. A list of devices appears. (You can view them in a list by selecting the **List** button or view details by selecting the **Details** button.)

   c. Select the desired device, and the click **Finish**. The new pilot appears in the field.

- **Prompt for Mailbox on Transfer**: Indicates whether a transfer announcement call should prompt a transferring party for a mailbox number. It is set to Yes by default, for SIP voice mail. For all other applications, it is set to No by default.

- **Validate Mailbox on Transfer**: Indicates whether the system validates the collected mailbox number on a transfer. It is set to No by default.

PROGRAMMING SIP PEER GROUP LISTS

The Group Lists folder provides a list of the group lists associated with the SIP peer. Group lists can be used by any subscriber for sending messages to several mailboxes simultaneously. At this time, the MiVoice Office 250 Call Processing only needs to know about group lists for OAI queries.

To create a Group List:

1. Right-click anywhere in the right side of the screen, and then select **Create Group List**. The Create SIP Group List Extension dialog box appears.

2. Type or select the Group List number, and then click **OK**.

3. Type or select the number in the Extension field. If you attempt to type an invalid number, a number that conflicts with an existing extension, or a number that is already assigned, a warning message appears, allowing you to fix the error. Type or select a new number, and then click **OK**.

4. Type a description of up to 20 characters in the Description text box. This description is used in the mailbox directory and should be entered in the form "last name, first name" (with a comma and space separating the names). Type a user name of up to 10 characters in the Username box. This is the name that appears on phone displays. Do not use slash (/), backslash (\), vertical slash (|), or tilde (~) characters in user names. Do not use Control characters in descriptions or user names.

5. Press **ENTER** or click on another field to save the change.

To delete a Group List:

Right-click the Group List, and then select **Delete**.

PROGRAMMING SIP PEER MAILBOXES

All mailboxes on the MiCollab Unified Messaging must be created as mailboxes on the MiVoice Office 250. This folder provides a list of the mailboxes associated with the SIP peer. The process for creating associated and non-associated mailboxes remains the same as the process for creating mailboxes for MiVoice Office 250 Voice Mails. At this time, the MiVoice Office 250 Call Processing only needs to know about associated/unassociated mailboxes for OAI queries/ commands. If the mailbox is associated, the Description and Username fields are read-only.

CONFIGURING SIP PEER PROGRAMMING OPTIONS

The following section describes the programming options that are available for SIP peers.

To configure SIP peer programming options, do one of the following:

- Select System – Devices and Feature Codes – SIP Peers – SIP Phone Groups –*<SIP phone group>* – **Configuration**.

- Select System – Devices and Feature Codes – SIP Peers – SIP Trunk Groups –*<SIP trunk group>* – **Configuration**.

- Select System – Devices and Feature Codes – SIP Peers – SIP Voice Mails –*<SIP voice mail>* – **Configuration**.

You can program the following SIP peer programming options.

Table 120: SIP Peer Programming Options

OPTION \ SIP PEER TYPE	REF. PAGE	SIP PHONE GROUPS	SIP TRUNK GROUPS	SIP VOICE MAILS
Registrar	718		✓	
Authentication	719	✓	✓	
MWI	719	✓		✓
Keep-Alive	720		✓	✓
NAT Settings	720	✓	✓	✓
Alternate IP/FQDN List	721		✓	
Route Sets	721		✓	✓
Registrations	722	✓		
IP Address	723	✓	✓	✓
Port Number	723	✓	✓	✓
Fully Qualified Domain Name	723	✓	✓	✓
Call Configuration	723	✓	✓	✓
Camp-Ons Allowed	724	✓	✓	✓
Operating State	724	✓	✓	✓
Maximum Number of Calls	725	✓	✓	✓
Supports Menu Softkeys	726			✓
Use ITU-T E.164 Phone Number	726		✓	
Static Binding	726	✓	✓	✓
Use Peer Address In From Header	726	✓	✓	✓

Page 1 of 2

Table 120:   SIP Peer Programming Options (continued)

OPTION \ SIP PEER TYPE	REF. PAGE	SIP PHONE GROUPS	SIP TRUNK GROUPS	SIP VOICE MAILS
Use Registered Username	**726**			✓
Disable Domain Validation	**727**	✓	✓	✓
DTMF Decoding Payload	**727**			
Supports Display Updates	**727**	✓		
Supports Ad Hoc Conferencing	**727**	✓		

Page 2 of 2

REGISTRAR

(This option is for SIP trunk groups only.) If the SIP peer does not require registration, the fields in this folder do not need to be configured. The Enable Registration option is set to No by default and the remaining fields appear with a red "X."

Configure the following options:

- **Enable Registration**: Set this to **Yes**, if the SIP peer requires registration. By default, this is set to No.

- **Registrar IP Address**: The SIP peer registrar server IP address. If this value is specified, the SIP REGISTER is sent to this address. The range is 0.0.0.0–255.255.255.255; the default is 5060. After you change this value to a valid IP address, you cannot change it back to the default value.

- **Registrar IP Port**: The SIP peer registrar server port number. The range is 0–65535.

- **Registrar FQDN**: The SIP peer registrar sever fully qualified domain name (FQDN). If this value is specified, the SIP REGISTER is sent to this address, but only if the Registrar IP address is set to 255.255.255.255. Enter a domain name, up to 254 characters, in the text box.

- **Registration Refresh Interval**: The registration refresh interval in seconds. When you create a SIP peer in the system that requires registration, the MiVoice Office 250 continues sending SIP REGISTER refresh requests to the SIP peer as per this interval. The range is 30–32767 seconds.

- **Initial Retry Interval**: This is the time in seconds that the MiVoice Office 250 should wait before retrying the registration with the SIP peer for the first time after time out. The next retry would be made after twice the initial delay, and so forth. The range is 0–32767 seconds.

- **Maximum Registration Retries**: This is the total number of retries before the MiVoice Office 250 stops retrying registration with the SIP peer. The range is 0–32767.

AUTHENTICATION

Configure the following options:

IMPORTANT	**Security Issue**: Failure to use a secure username and password could result in the MiVoice Office 250 being compromised. If **Enable In-bound Authentication** is set to **Yes**, the SIP Phone can only register with a username and password in the request. The preference is to use the **In-bound Authentication Username** and **In-bound Authentication Password**. If both the **In-bound Authentication Username** and **In-bound Authentication Password** are empty, the SIP Phone can register either using the extension number or the phone passcode in the request. If **Enable In-bound Authentication** is set to **No**, the SIP Phone can register using just the extension number.

- **Enable In-Bound Authentication**: If the Enable In-Bound Authentication flag is enabled for a SIP peer, incoming calls and SIP requests from the SIP peer are authenticated by the MiVoice Office 250.

- **In-bound Authentication Username**: When in-bound authentication is enabled, the SIP peer uses this username as part of the credentials while authenticating itself with the MiVoice Office 250. The username can contain up to 64 characters. This credential applies to all SIP phones under this group. This field is used mostly for SIP phones. However, a SIP Trunk Group may use these fields if needed.

- **Inbound Authentication Password**: When in-bound authentication is enabled, the SIP peer uses this password as part of the credentials while authenticating itself with the MiVoice Office 250. The password can contain up to 64 characters. This credential applies to all SIP phones under this group. This field is used mostly for SIP phones. However, a SIP Trunk Group may use these fields if needed.

NOTE	In addition to the two SIP Phone Group-level credentials described above (In-bound Authentication Username and Password), the MiVoice Office 250 also uses SIP Phone-level credentials in SIP phones. They are specified by the Extension and Password fields in System\Devices and Feature Codes\Phones\<*SIP phone*>.

- **Out-bound Username**: This field applies only if the SIP peer requires registration or call authentication. The username is provided by the SIP peer. The MiVoice Office 250 uses this username as part of the SIP Digest Credentials in the SIP Request. The username is limited to 64 characters.

- **Out-bound Password**: This field applies only if the SIP peer requires registration or call authentication. The password is provided by the SIP peer. The password is limited to 64 characters.

- **Do not use Out-bound Username in REGISTER**: This flag controls the registration of a SIP call when upgrading from Release 6.0.

MWI

The Message Waiting Indication (MWI) field determines whether the system accepts the MWI from the SIP peer. Verify that the **Accept MWI** option is set to **Yes**. To have the system ignore MWI from the SIP peer, change the setting to **No**. It is set to Yes by default.

For SIP phones, the MWI functionality works only if the SIP phone is statically registered.

KEEP-ALIVE

The SIP peer has keep alive functionality to refresh the NAT bindings for any firewall/NAT in the path and to determine if the SIP peer is reachable or not. You can enable or disable the SIP peer keep alive functionality in DB Programming for each SIP peer. When you enable the keep alive functionality, the MiVoice Office 250 sends a SIP OPTIONS request to ping the SIP peer. When you disable the keep alive functionality, the MiVoice Office 250 does not send any SIP OPTIONS requests to the SIP peer.

The following options are available:

- **Enable Pinging**: If set to Yes, the system starts sending SIP OPTIONS request towards the SIP peer for pinging purposes.

- **Ping Interval**: Indicates the number of seconds that the system waits to ping the SIP peer. The range is 30–32767 seconds.

- **Ping Failure Threshold**: Indicates the maximum number of consecutive ping failures that would put this SIP peer Out-Of–Service. The range is 0–255.

If the SIP OPTIONS requests fail consecutively for the configured number of times, the corresponding SIP peer is put to Out-Of-Service state and the following system alarm is displayed: ALARM #145 SIP peer Out-of-Service.

On the success of any subsequent SIP OPTIONS request, the SIP peer is put back to In-Service (INS) state from Out-Of-Service (OOS) state. This would also clear the previously raised system alarm.

NAT SETTINGS

The NAT Settings option specifies the NAT address type (refer to the *MiVoice Office 250 Installation Manual* for details about programming this option).

For SIP Trunk Groups:

The default is "No NAT or SIP-Aware NAT" (for systems that are using a SIP-aware firewall). If you are not using a SIP-aware firewall, you must change the setting to "Non SIP-Aware NAT". There are two choices:

- **No NAT or SIP-Aware NAT**: Used in two scenarios: When the MiVoice Office 250 is directly in the public network, without a firewall or NAT device between the MiVoice Office 250 and the SIP peer, and when the firewall between the MiVoice Office 250 and the SIP trunk service provider is SIP-aware. Refer to the *MiVoice Office 250 Installation Manual* for a list of the tested firewalls.

- **Non SIP-Aware NAT**: Used when the firewall between the MiVoice Office 250 and the SIP trunk service provider is not SIP-aware.

For SIP Phone Groups and SIP Voice Mails:

There are two choices:

- **Native**: Used for internal SIP phones (those that do not pass through near-end NAT).

- **NAT**: Used for external SIP phones (those that do pass through near-end NAT)

ALTERNATE IP/FQDN LIST

(This option is for SIP trunk groups only.) Some providers use multiple IP addresses to send SIP messages to the MiVoice Office 250. You must add All IP addresses or FQDNs other than the primary IP/FQDN to the list for all calls to be successful.

To add alternate IP addresses or fully qualified domain names:

1. Right-click the right pane in the Configuration folder.

2. Select Add to **Alternate IP/FQDN List**. The following dialog box appears.
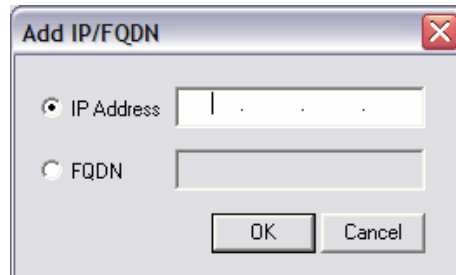
3. Click the **IP Address** option, and then type an IP address or click the **FQDN** option, and then type a fully qualified domain name. An error message appears if you attempt to create an IP address of fully qualified domain name that already exists.

ROUTE SETS

SIP Route Set is a collection of ordered SIP URLs which represent a list of SIP Proxy Servers that must be traversed while sending a particular SIP request to the corresponding SIP peer. This field is mainly used for specifying an outbound SIP Proxy Server that must be used to send SIP requests to the SIP peer. To set an outbound SIP Proxy Server, Mitel recommends that only one route set element (for example, SIP URL) be added in this list.

The following components of a Route Set element comprise the SIP URL:

*   **Username**: Indicates the user information part of the SIP URL.

*   **Hostname**: Indicates the host part of the SIP URL. This field can be a FQDN (Fully Qualified Domain Name) or the IP address of the SIP Proxy Server.

*   **Port Number**: Indicates the port number of the SIP URL. This field specifies the listen port of the SIP Proxy Server.

*   **Transport**: Indicates the transport parameter of the SIP URL. This field specifies the transport protocol that should be used while sending SIP requests towards the SIP Proxy Server. However, at this point only UDP transport protocol is supported and this field is read-only.

*   **URL Scheme**: Indicates the URL scheme of the SIP URL. The possible values are SIP and SIPS (for secure transport). However, at this point only SIP URL scheme is supported and this field is read-only.

*   **URL Parameters**: Indicates SIP URL parameters. If the SIP Proxy Server expects any URL parameters, those parameters (for example, "param1=value1;param2=value2") can be specified in this field.

- **Loose Router**: Indicates whether the SIP Proxy Server is a loose router as specified by RFC 3261. If this flag is enabled, it adds a URL parameter flag "lr" in the corresponding SIP URL.

For example, to specify an outbound SIP Proxy Server (that is listening on UDP port 5060 and IP Address 192.168.0.1, and is a loose router), you must add one route set element with the following information:

- Username: *<blank>*

- Hostname: 192.168.0.1

- Port Number: 5060

- Transport: UDP

- URL Scheme: SIP

- URL Parameters: *<blank>*

- Loose Router: Yes

REGISTRATIONS

(This option is for SIP phones only.) You can register SIP phones with the MiVoice Office 250 dynamically or statically.

- For dynamic registration, the status of a SIP phone is determined by the existence of an active registration in the system for that SIP phone. When a SIP phone registers with the system, its status becomes "Idle" (online) as long as there is a valid SIP phone (Category F Phones) license available and becomes "Offline" when the registration expires or SIP phone un-registers.

- For static registration, the status of a SIP phone becomes "Idle" (online) as long as there is a valid SIP phone (Category F Phones) license available.

This folder allows you to configure the following settings that are required for registration per-SIP Phone Group basis:

- **Address of Record**: Indicates the Address of Record that the SIP peer uses to register with the MiVoice Office 250. This field is for read-only.

- **Registration URI**: Indicates the SIP URI representing the Contact address in the SIP REGISTER request from the SIP peer that created this dynamic binding. This field is for read-only.

- **Registration Call ID**: Indicates the SIP Call ID of the SIP REGISTER request received from the SIP peer that created this dynamic binding. This field is for read-only.

- **Registration Cseq Number**: Indicates the SIP Cseq number of the SIP REGISTER request received from the SIP peer that created this dynamic binding. This field is for read-only.

- **Registration Update Time**: Indicates the timestamp when the SIP REGISTER request was received from the SIP peer and updated the dynamic binding. This field is for read-only.

- **Registration Expire Time**: Indicates the time in seconds to expire this registration since it was last updated. This field is for read-only.

IP ADDRESS

The IP Address option indicates the IP address of the SIP peer. MiVoice Office 250 uses this IP address as the destination address of the SIP peer to send SIP Messages. If this IP address is not specified (for example, set to 255.255.255.255), MiVoice Office 250 uses the FQDN (Fully Qualified Domain Name) field of the SIP peer as the destination address to send SIP Messages.

For SIP Trunk Groups, this is provided by the SIP trunk service provider.

For SIP Phone Groups, this is provided by the site administrator who specifies the IP address of the SIP phones. Also, this is used only in case the SIP phones do not register with the system and the Static Binding field is set to true.

For SIP Voice Mails, this is provided by the site administrator who specifies the IP address of the SIP Voice Mails.

PORT NUMBER

The Port Number option indicates the port that the system listens on the system for SIP peer messages. The range is 0–65535; the default is 5060 (standard SIP port).

NOTE	For MiVoice Office 250 systems configured with MiCollab Unified Messaging Release 15 and up, program this port setting to 5058.

FULLY QUALIFIED DOMAIN NAME

Indicates the fully qualified domain name (for example, sip-service.mitel.com) of the SIP peer. This field is optional and used only if the IP Address is not specified (that is set to 255.255.255.255).

Enter an FQDN, up to 254 characters, in the text box. The default is blank.

For SIP Trunk Groups, this is provided by the SIP trunk service provider.

For SIP Phone Groups, this is provided by the site administrator who specifies the FQDN of the SIP phones. Also, this is used only in case the SIP phones do not register with the system and the Static Binding field is set to true.

For SIP Voice Mails, this is provided by the site administrator who specifies the FQDN of the SIP Voice Mails.

CALL CONFIGURATION

Clicking **Call Configuration** takes you to the Call Configuration folder (System\IP-Related Information\Call Configurations\<*call configuration number*>). When you create a SIP peer without using a template, by default the new SIP peer is added to Call Configuration 1 <*Local*>. For other SIP peer trunk groups, by default the number is assigned sequentially to the next available number. See the following example.
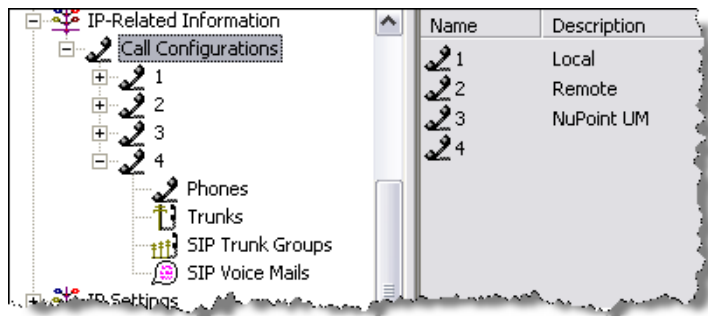
Refer to the *MiVoice Office 250 DB Programming Help* for details about each field and their default settings. See "IP Call Configurations" on **page 629** for information about fields in the Call Configuration folder.

CAMP-ONS ALLOWED

The Camp-Ons Allowed option enables or disables the Camp-On feature for each SIP peer. If disabled, users placing outgoing calls hear busy signals when all SIP peers in the group are in use or unavailable. If enabled, users can camp on and wait for an available SIP peer. It is set to Yes by default.

OPERATING STATE

The Operating State option indicates the operating state of the SIP peer. The SIP peer operating states are:

- **In Service (INS)**: The SIP peer is available and in service. This is the default setting.

- **Out–of–Service (OOS)**: The SIP peer is temporarily unavailable. The SIP peer reaches this state due to one of the following reasons:

  - The outgoing call attempts failed consecutively for a configured number (Call Failure Threshold) of times.

  - The ping requests (SIP OPTIONS) timed out or failed for a configured number (Ping Failure Threshold) of times.

  - Alarm 145 (SIP Peer Out-of-Service) generates when a SIP peer state changes from INS to OOS. This alarm clears automatically when the SIP peer state changes from OOS or OOS–Maint to INS.

- **Out–of–Service–Maintenance (OOS–Maint)**: The administrator manually put the SIP peer out of service to do maintenance. When an administrator changes Operating State of a SIP peer to "Out-Of-Service-Maintenance," the SIP peer status becomes "Offline."

SIP Peer Status

A SIP peer reports its status to any monitoring device. Each status includes the light-emitting diodes (LED) flash rate associated with a DSS/BLF (PKM) button programmed for the corresponding SIP peer. The SIP peer statuses are:

- **Idle**: The operating state is INS and there is at least one available trunk or port for placing an outgoing or incoming call. The flash rate is off.

- **Busy**: The operating state is INS and there are no available trunks or ports for placing an outgoing or incoming call. The flash rate is solid.

- **Offline**: The operating state is Out-of-Service (OOS) or OOS-Maintenance (OOS-Maint). The flash rate is solid.

MAXIMUM NUMBER OF CALLS

The functionality of the Maximum Number of Calls field varies depending on the SIP peer type.

For SIP Phone Groups

Each individual SIP phone under a SIP Phone Group can only be on two concurrent connected, holding, or ringing calls at one time. If the SIP phone attempts to place a third call, the MiVoice Office 250 rejects the call. For calls to the SIP phone, the system uses the "Maximum Number of Calls" to determine how many concurrent connected, holding, or ringing calls the SIP phone can support.   The value of this field can only be a one or a two.

Some SIP phones support call-waiting, which is the ability to accept a second call when the SIP phone is connected on a call. When the SIP phone receives the second call, it notifies the user that a second call is ringing in. To enable this functionality, set the "Maximum Number of Calls" to two. If the SIP phone does not support call-waiting, set the "Maximum Number of Calls" to one. If the number of concurrent calls equals the configured value, calls to the SIP phone camp-on. Calls that are camped on remain camped-on until the number of calls falls below the threshold. Note that if the "Maximum Number of Calls" is set to a value other than a one or two, it will use a value of one.

The UC Express Softphone does not support call-waiting so this field needs to be set to one.

The 5610 Cordless Handset, and the Mitel 5603/04/07 SIP Phones can be configured to support call-waiting. If the SIP phone is set to support call-waiting, set the "Maximum Number of Calls" to two; otherwise, set the field to one.

When programming a Mitel UC 360 Collaboration Point, set this field to 4.

For SIP Peer Trunk Groups

Indicates the maximum number of concurrent calls that are permitted towards the SIP peer trunk groups. DB Programming restricts this field based on the number of the SIP Trunks licenses. The range is 0 - 32767; the default value is based on the number of trunks and licenses. This value increments when you add a SIP peer trunk and this value decrements when you delete a SIP peer trunk. The value must be less or equal to the number of SIP Trunks licenses for the system.

For SIP Peer Voice Mails

Indicates the number of available SIP Voice Mail ports. DB Programming restricts this field based on the number of the SIP Voice Mail Ports licenses. The range is 0-32757 ports; the default is 0. The value must be less or equal to the SIP Voice Mail Ports licenses the system has.

SUPPORTS MENU SOFTKEYS

(For SIP voice mails only.) The Supports Displays option indicates whether the SIP Peer Voice Mail provides display menus and soft keys that should be displayed on the end points. It is set to **Yes** by default.

USE ITU-T E.164 PHONE NUMBER

The Use ITU-T E.164 Phone Number option indicates if the SIP peer requires MiVoice Office 250 to use ITU-T E.164 formatted phone numbers while sending SIP requests towards the SIP peer. If set to Yes, the MiVoice Office 250 uses ITU-T E.164 formatted phone numbers while sending SIP INVITE messages towards the SIP peer. For incoming SIP requests from the SIP peer, ITU-T E.164 formatted phone numbers are always supported irrespective of this flag.

STATIC BINDING

The Static Binding option specifies whether a static binding exists for the corresponding SIP peer. If set to Yes, the IP address and/or hostname, and Listening Port number must be set for the corresponding SIP peer. It is set to Yes for SIP Trunk Group and SIP Voice Mail and No for SIP Phone Groups.

USE PEER ADDRESS IN FROM HEADER

The Use Peer Address In From Header option specifies whether a SIP peer address (IP address or FQDN) is used in the "From" header of SIP requests towards a SIP peer. Essentially, the same SIP peer address is used in the "From" and "To" headers of SIP requests sent towards a SIP peer. Note that for SIP REGISTER requests towards a SIP peer, the same peer address is used in the "From" and "To" headers regardless of this flag. It is set to No by default.

USE REGISTERED USERNAME

The Use Registered Username option specifies whether a SIP peer should use the registered username for the user.

When the MiVoice Office 250 sends an INVITE messaged to a SIP device through the Mitel Border Gateway (MBG), the MBG expects the SIP URI to contain the username of the SIP device that the MBG used when it registered the SIP device with the PBX.

When this flag is set to Yes, the MiVoice Office 250 will use the registered username instead of the called-party number for the username component of the SIP URI. Set this flag to Yes when the MiVoice Office 250 is connected to an MBG.

When this flag is set to No (default setting), the MiVoice Office 250 will use the called-party number.

DISABLE DOMAIN VALIDATION

This field indicates if the domain validation is disabled for the corresponding SIP Peer (Phone Group, Trunk Group, Voice Mail). Select **Yes** or **No** from the drop-down list.

If enabled (**Yes** is selected), then MiVoice Office 250 would not validate the domain or IP address in the To and/or Request URI of the incoming SIP Requests from the corresponding SIP Peer.

DTMF DECODING PAYLOAD

SIP Peers in Database Programming can be configured with the necessary custom payload type value, depending on the supported payload type of the far end SIP provider. In previous releases, this value was configured in IP Settings. The payload type is typically negotiated between carriers and devices, but in some cases the SIP service provider does not accept requests. The DTMF Payload Type is made available by the SIP trunk service provider.

The default DTMF Decoding Payload value applies to North American, UK, and Australian systems.

SUPPORTS DISPLAY UPDATES

This field indicates if phones belonging to a particular SIP Phone Group support display updates. Select either **Yes** or **No** from the drop-down list.

The display updates include the following functionality:

- Outbound calls

- Forwards

- Transfers

- Conferencing

This functionality is geared for a 5603/5604/5607 SIP phone; however, if the SIP phone supports this functionality it can be used.

SUPPORTS AD HOC CONFERENCING

This field indicates if phones belonging to a particular SIP Phone Group support ad-hoc conferencing. Select **Yes** or **No** from the drop-down list.

This functionality is geared for a 5603/5604/5607/5610 SIP phone; however, if the SIP phone supports this functionality it can be used.

SIP PEER DIAGNOSTICS

This section describes diagnostics applications that are available for SIP peers.

SIP VIEW DIAGNOSTICS FEATURE CODE

The Phone Administrator can use the existing diagnostics feature code "Diagnostics – SIP View" to view the incoming and outgoing SIP peer messages. This feature exists today for SIP trunks. This diagnostic code changes the SIP values system-wide which determines the output that appears in Message Print and diagnostic log files.

SIP PEER DUMPS IN SYSTEM MONITOR

The following dumps are used for SIP peers:

- **Dump SIP Dialog Manager**: Contains information and statistics about SIP dialogs. You can also get these statistics from the AWP System SIP Diagnostics Web page.

- **Dump SID**: Contains dynamic information about a particular active SIP call. The extension number for each SIP call is a random non-dialable extension that you obtain by first performing a Dump Call Manager during the active call. After you obtain the extension for the current call, you can perform Dump Extension to get the SID information.

- **Dump SIP Peer**: Contains information about the SIP peer, such as the IP address, UDP port number, and host name.

The Phone Administrator can use the existing diagnostics feature code "Diagnostics – Dump Extension" to perform these dumps.

SIP LOG FILES DIAGNOSTICS

The Log Files page includes diagnostic log files for SIP peers: CP SIP and CP History. The CP SIP log file includes SIP message and SIP peer information. The CP History log file includes SIP peer message information depending on the SIP View option (see **SIP View Diagnostics Feature Code** above. You can view the logs in the AWP. Refer to the AWP Help for details.

Chapter 12

SYSTEM SETTINGS

INTRODUCTION

This chapter describes features that you can use to customize your system settings:

For troubleshooting tips for system-level issues, see **page 781**.

SYSTEM-WIDE PARAMETERS

You can program system-wide parameters, which include date, time, and language options, as shown in the figure below:
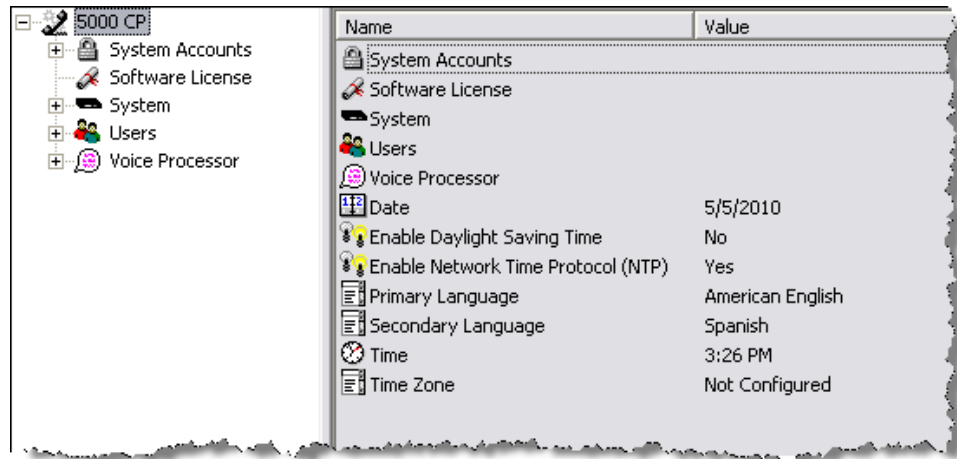
This section contains the following programming instructions:

- **Setting the System Date** below

- Setting the System Time on **page 732**

- Selecting the System Time Zone on **page 732**

- Programming Primary and Secondary Languages on **page 732**

- Programming Daylight Saving Time [British Summer Time] on **page 733**

SETTING THE SYSTEM DATE

| NOTE | If the Enable Network Time Protocol (NTP) option is enabled, the system date is controlled by the NTP server and the Date field is read-only. |

To program the system date:

1. Select **Date**.

2. In the **Value** column, type the date or select the date from the calendar.

3. Click out of the field or press **ENTER** to save your change.

SETTING THE SYSTEM TIME

> **NOTE** If the Enable Network Time Protocol (NTP) option is enabled, the system time is controlled by the NTP server and the Time field is read-only.

To change the time:

1. Select **Time**.

2. In the **Value** column, type the time or select the time from the list.

3. Click out of the field or press **ENTER** to save your change.

SELECTING THE SYSTEM TIME ZONE

You can select the time zone for your system.

To change the time zone:

1. Select **Time Zone**.

2. In the **Value** column, select the time zone from the list.

3. Click out of the field or press **ENTER** to save your change.

PROGRAMMING PRIMARY AND SECONDARY LANGUAGES

The System provides a choice between American English, British English, Spanish, Canadian French, and Japanese prompts and displays. The system selects the language to use for each call, as determined by the trunk, phone, and voice processor programming.

> **NOTES** Japanese prompts can be viewed only on phones with an LCD display (excluding the 8690). The Japanese language is not supported on 52xx/53xx-series IP phones.

If you are using a Unified Voice Messaging (UVM) voice processing system, make sure that the appropriate prompts have been loaded into the voice processor. Otherwise, if users try to access the prompts, they hear the default American or British English prompts. For more information about loading voice processor prompts, refer to the Installation chapter in the *MiVoice Office 250 Installation Manual* .

For complete information about multilingual capability, see "Multilingual Capability" on .

To select the primary and secondary languages:

1. Select **Primary** (or **Secondary**) **Language**.

2. In the **Value** column, select the language from the list.

3. Click out of the field or press **ENTER** to save your change.

PROGRAMMING DAYLIGHT SAVING TIME [BRITISH SUMMER TIME]

This section contains the following information:

- **Feature Description** below

- Programming Daylight Saving Time on **page 734**

FEATURE DESCRIPTION

The Daylight Saving Time option determines whether the system automatically adjusts the time when daylight-saving time [British summer time] occurs. If enabled, standard time changes on the days specified. The system time is also modified when DST ends. If disabled, the system does not recognize daylight-saving time [British summer time]. In the default state, the DST flag is disabled.

> **NOTICE**
>
> Station Message Detail Recording (SMDR) generates call costs based on the difference between the start and stop times of a call. System time changes *will* affect this calculation. If Daylight Saving Time is enabled for the system, and the time changes while SMDR is tracking a call, the call cost will be inaccurate. For example, if a call starts at 1:30 AM and ends at 2:30 AM on the night that Daylight Saving Time goes into effect, the call cost will be for 2 hours (1:30 to 3:30) instead of 1 hour (1:30 to 2:30).

System Time Adjustments

You can program the system so that it automatically adjusts the system time whenever DST is or is not in effect. If properly programmed, the system will adjust the time as follows:

- *On the appropriate day in spring (USA only),* the system time will "spring ahead" one hour at 1:59:59 AM, changing the time to 3:00:00 AM.

- *On the appropriate day in spring (Europe only),* the system time will "spring ahead" one hour at 12:59:59 AM, changing the time to 2:00:00 AM.

- *On the appropriate day in fall (USA and Europe),* the system time will "fall back" one hour at 1:59:59 AM, changing the time to 1:00:00 AM.

When the time changes, the system issues a message to Message Print. For details, refer to the *Message Print Diagnostics Manual*, part no. 550.8018.

Interaction with SMDR

Station Message Detail Recording (SMDR) generates call costs based on the difference between the start and stop times of a call. System time changes affect this calculation. If DST is enabled and the time changes while SMDR is tracking a call, the call cost will be inaccurate. For example, if a call starts at 1:30 AM and ends at 2:30 AM on the night that DST goes into effect, the call cost will register as two hours (1:30–3:30) instead of one hour (1:30–2:30).

Interaction with Scheduled Resets

Because this feature changes the system time, a scheduled delayed major reset may be affected as follows:

- *If the reset is scheduled to occur within four minutes of the time change,* the reset is delayed by five minutes. For example, if the reset is scheduled for 1:58, the reset will occur at 2:03, which is five minutes later.

- *If the reset is scheduled to occur between 2:00:00 AM and 3:00:00 AM [1:00:00 to 2:00:00 in Europe] in the spring,* the system does *not* reset. Instead, the system performs a delayed major at the next scheduled date and time.

- *If the reset is scheduled to occur between 1:00:00 AM and 2:00:00 AM in the fall,* the system resets twice—once before the time change and once after. For example, if the reset time is set for 1:30 AM on the day of the fall time change, the system resets at 1:30 AM. Then, at 2:00 AM (1:59:59), the time changes to 1:00 AM. When it is 1:30 AM, the system resets again.

NOTE	Mitel recommends that you *not* schedule automatic functions, such as backups and resets, to occur at 2:00 AM (1:00 AM or 2:00 AM in Europe). If automatic functions are scheduled at these times, the system may not perform the function when the time changes.

Because a major system reset is not required for these actions to take effect, Administrators are not prompted to perform an immediate or scheduled reset if the changes are programmed from an administrator's phone. Rather, the database changes take effect immediately.

PROGRAMMING DAYLIGHT SAVING TIME

To turn the Daylight Saving Time flag on:

1. Select **Enable Daylight Saving Time**.

2. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

3. Click out of the field or press **ENTER** to save your change.

ECHO PROFILES

All system devices are associated with an echo profile. Besides voice mail and conferencing, echo profiles also apply to physical devices of the system, including:

- Phones/Connections

- Trunks

- Span Echo Profiles - Dual T1/E1/PRI Devices

You can program Echo Profiles for trunks (see the following section) or individual phones (see **page 417**).

PROGRAMMING ECHO PROFILES FOR TRUNKS

The following are default Echo Profile options for trunks:

- No Echo (Disabled)

- Low Echo - no NLP (Low Aggressiveness, no Non-Linear Processing)

- Low Echo (Low Aggressiveness)

- Medium Echo (Medium Aggressiveness), and

- High Echo (High Aggressiveness).

Devices programmed for the system are automatically assigned a particular default configuration according to their type (see **page 738** for the default echo profiles for devices).

Table 121:   Trunk Echo Profiles

DB PROGRAMMING FIELD	DEVICE TYPE	DEFAULT ECHO PROFILE	DEFAULT SPAN ECHO PROFILE FOR DUAL T1/E1/ PRI DEVICES
System\Devices and Feature Codes\Phones	Digital Telephones	LOW ECHO	N/A
	IP Phones (IPSLA, IP Softphone, IP Endpoint)	NO ECHO	N/A
	Single Lines	LOW ECHO	N/A
	OPXs programmed on single T1/E1/PRI Module	LOW ECHO	N/A
	OPXs programmed on a Dual T1/E1/PRI Module	NO ECHO	LOW ECHO
System\Devices and Feature Codes\IP Connections	IP Connections (IP Networking)	NO ECHO	N/A
System\Devices and Feature Codes\Modems	Modems	NO ECHO	N/A

Table 121:   Trunk Echo Profiles (continued)

DB PROGRAMMING FIELD	DEVICE TYPE	DEFAULT ECHO PROFILE	DEFAULT SPAN ECHO PROFILE FOR DUAL T1/E1/ PRI DEVICES
System\Devices and Feature Codes\Trunks	Loop Start Trunks (analog)	MEDIUM ECHO	N/A
	IP Trunks (SIP, MGCP)	NO ECHO	N/A
	Single T1/E1/PRI CO Trunks (Loop Starts, Ground Starts[1], DIDs, E&Ms, B-Channels on a port *not* programmed for private networking)	MEDIUM ECHO	N/A
	Single T1/E1/PRI B-Channel Trunks programmed on a port using private networking	MEDIUM ECHO	N/A
	Dual T1/E1/PRI CO Trunks (Loop Starts, Ground Starts, DIDs, E&Ms, B-Channels on a port *not* programmed for private networking)[2]	NO ECHO	MEDIUM ECHO
	Dual T1/E1/PRI B-Channel Trunks programmed on a port using private networking	NO ECHO	MEDIUM ECHO
	DID Trunk (on a DEM-16)	LOW ECHO	N/A
	Basic Rate Trunk	LOW ECHO	N/A

Page 2 of 2

1. T1 ground start trunks do not support Mid Call Features (MCF), and Caller ID.
2. A B-Channel trunk programmed for private networking is automatically moved into the correct echo profile by the system, so their echo profiles are read-only in DB Programming.

The following two fields set an echo profile for two main system functions:

- Voice Processor Echo Profile (the default is set to No Echo)

- Conferencing Echo Profile (the default is set to Medium Echo)

These fields apply their associated echo profile settings to any of their functions for the whole system.

To change the echo profile for a trunk:

1. Select System – Trunks – **Echo Profiles**.

2. In the **Value** column, right-click, and then select **Change Echo Profile**. A wizard appears allowing you to select any of the echo profiles for this field.

To move a particular device from one echo profile to another:

Do one of the following:

- Select the device from the current profile and drag/drop it onto the desired configuration in the tree view.

- Use the context menu option of the desired profile's folder to move the desired devices. To the right is an example of the move options of the Phones/Connections folder's context menu.

PROGRAMMING ECHO PROFILES FOR PHONES

The following devices are associated with an Echo Profile by default: all digital telephones, single lines, IP Phones, IP Softphones, IPSLAs, Modems, IP Connections, and trunks. For more information about programming Echo Profiles for system trunks, see **page 735**. Below is an example of the Echo Profile fields for a phone.

Each echo profile has the following three subfolders:

- Phones/Connections

- Trunks

- Span Echo Profiles - Dual T1/E1/PRI Devices: Span echo profile has the following subfolders:

  - Phones

  - Trunks

The devices in the first two subfolders use their associated echo profile for echo generated by the MiVoice Office 250. The span-side devices cancel echo generated outside the system that comes in through one of the ports on the Dual T1/E1/PRI Module. Therefore, while all physical devices on the system have an echo profile, only devices programmed on a port of a Dual T1/E1/PRI Module have a span-side echo profile.

To add a device to the list:

1. Select System – Devices and Feature Codes – **Phones**.

2. Double-click **Echo Profiles**.

3. Double-click the echo profile type.

4. Right-click the right pane, and then select **Move To List**. A wizard appears allowing you to select any of the devices for this field.

5. Click **Next**.

6. Select the devices, and then click **Move Items**.

7. Click **Finish**.

To move a particular device from one echo profile to another:

Do one of the following:

- Select the device from the current profile and drag/drop it onto the desired configuration in the tree view.

- Use the context menu option of the desired profile's folder to move the desired devices.

The default echo profile for each device appears in **Table 122**.

Table 122:   Default Echo Profiles for Devices

DB PROGRAMMING FIELD	DEVICE TYPE	DEFAULT ECHO PROFILE	DEFAULT SPAN ECHO PROFILE - DUAL T1/E1/PRI DEVICES
System\Devices and Feature Codes\Phones	Digital Telephones	LOW ECHO	N/A
	IP Phones (IPSLA, IP Softphone, IP Phone)	NO ECHO	N/A
	Single Lines	LOW ECHO	N/A
	OPXs programmed on a single T1/E1/PRI Module	LOW ECHO	N/A
	OPXs programmed on a Dual T1/E1/PRI Module	NO ECHO	LOW ECHO
System\Devices and Feature Codes\IP Connections	IP Connections (IP Networking)	NO ECHO	N/A
System\Devices and Feature Codes\Modems	Modems	NO ECHO	N/A
System\Devices and Feature Codes\Trunks	Loop Start Trunks (analog)	MEDIUM ECHO	N/A
	IP Trunks (SIP, MGCP)	NO ECHO	N/A
	Single T1/E1/PRI CO Trunks (Loop Starts, Ground Starts[1], DIDs, E&Ms, B-Channels on a port NOT programmed for private networking)	LOW ECHO	N/A
	Single T1/E1/PRI B-Channel Trunks programmed on a port using private networking	MEDIUM ECHO	N/A
	Dual T1/E1/PRI CO Trunks (Loop Starts, Ground Starts, DIDs, E&Ms, B-Channels on a port NOT programmed for private networking)[2]	NO ECHO	LOW ECHO
	Dual T1/E1/PRI B-Channel Trunks programmed on a port using private networking	N/A	MEDIUM ECHO
	DID Trunk (on a DEM-16)	LOW ECHO	N/A
	Basic Rate Trunk	LOW ECHO	N/A

1. T1 ground start trunks do not support Mid Call Features (MCF), and Caller ID.
2. A B-Channel trunk programmed for private networking will automatically be moved into the correct echo profile by the system, so their echo profiles are read-only in DB Programming.

For each device, the associated echo profile is displayed. Double-clicking on the echo profile field changes the view to the folder of that echo profile.

To change the echo profile for a device:

1. Right-click on the device and select **Change Echo Profile**. A wizard appears allowing you to select any of the echo profiles for this field.

2. Click **Next**.

3. Select the echo profile, and then click **Finish** to save the change.

PROGRAMMING VOICE OVER INTERNET PROTOCOL (VOIP) ECHO CANCELLER

You can use the (improved) VoIP Echo Canceller, as shown in the figure below, to select the system-wide echo cancellation settings based on the amount of echo that your system is experiencing. Systems upgrading to version 3.0 or later maintain the "Basic" level of echo cancellation. New system installs default to the Advanced level.

For VoIP Echo Canceller troubleshooting, see **page 740**.
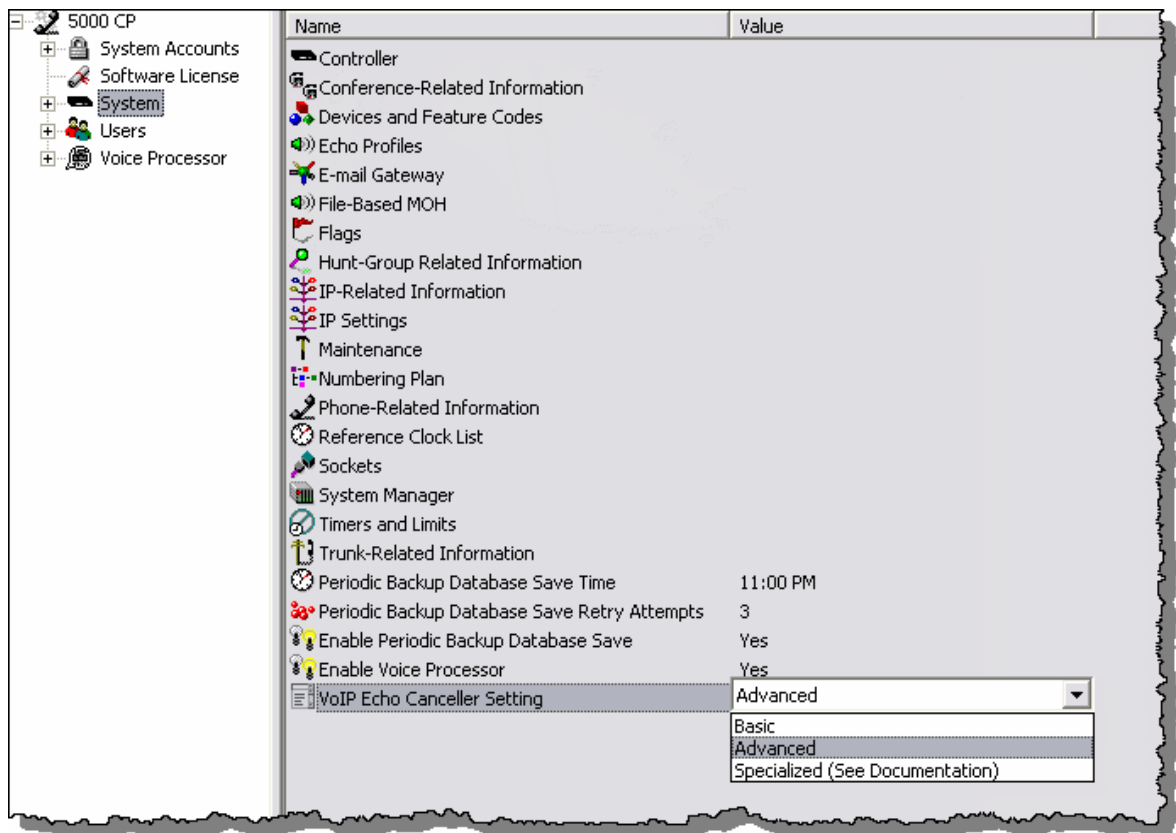
The following are VoIP Echo Canceller settings:

• **Basic**: Provides less echo cancellation, but allows more concurrent IP calls.

• **Advanced** (default): Provides the best echo cancellation for most customers but allows fewer concurrent IP calls than the Basic setting.

• **Specialized**: *Do **not** use this option unless you are instructed to do so by Mitel Technical Support. This option is only for sites that are experiencing extreme amounts of echo.* Allows the fewest amount of concurrent IP calls.

To program the VoIP Echo Canceller:

> **NOTE** The system performs a reset immediately after the VoIP Echo Canceller setting is changed.

1. Select System – **VoIP Echo Canceller Setting**.

2. From the **Value** column, select one of the options previously described.

3. Click out of the field or press **ENTER** to save the change.

Troubleshooting the VoIP Echo Canceller

Table 123 lists VoIP Echo Canceller troubleshooting scenarios.

Table 123: VoIP Echo Canceller Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
A customer complains about echo.	The system may still be configured to use the Basic VoIP Echo Canceller setting.	Configure the system to use the Advanced VoIP Echo Canceller setting and monitor the site for feedback.
A customer is using the Advanced VoIP Echo Canceller setting and complains about loud, clear echo throughout calls.	The echo has a delay of more than 64 ms.	Configure the system to use the Specialized VoIP Echo Canceller setting and monitor the site for feedback. Determine the source of the long echo, because this does not occur regularly with trunks that are operating normally.

FILE-BASED MUSIC-ON-HOLD (MOH)

This section contains the following information:

- **Feature Description** below

- Programming File-Based MOH Sources on **page 742**

- Troubleshooting File-Based MOH on **page 746**

FEATURE DESCRIPTION

The File-Based Music-On-Hold (MOH) feature expands the existing MOH source beyond the built-in audio port located on the back of the MiVoice Office 250 chassis. That is, you do not have to connect to an external music source. This feature uses the compact flash-type memory card to store MOH audio files. This feature requires a software license (see "System Software Licenses" on **page 49**).

| NOTE | A File-based MOH file cannot be used for background music. The Background Music feature (see **page 322**) always uses the physical MOH connection and not File-based MOH. |

Although a device may connect to a music source for a length of time longer than the length in time of the audio file associated with the music source, the file-based music source continuously loops the audio playback so that there is always audio output from the source. A default audio file (default_moh.n64u) is provided on the MiVoice Office 250 compact flash-type memory card. You can use this as a sample file to associate a music source after you configure it in v3.0 or later DB Programming. The sample file plays this message:

> "Mitel Network Communications Solutions enable organizations to blend their voice system into their data network, creating a cost-effective, efficient communications environment for small to medium businesses."

This feature supports the non-proprietary G.711 (.n64u) file format (8 KB per second). A File-Based MOH file uses 0.5 MB per minute of storage and stores up to 80% of the capacity of the compact flash type card used. Each File-Based MOH file can be of any size as long as the total of all File-Based MOH files does not exceed 80% of the compact flash size.

You can use the MOH Converter Utility to convert audio files into the proper format. TheMiVoice Office 250 MOH Utility uses the Sound eXchange (SoX) audio processing utility to convert the audio files to the desired format. When you install theMiVoice Office 250 MOH Utility with DB Programming, a folder called MOH Converter is installed in the same location. This folder contains various SoX text (.txt) and Portable Document Format (.pdf) files. Refer to the SoX .txt and .pdf files for additional information. You may also go to http://sox.sourceforge.net for more information. Supported audio file formats are listed in **Table 124**.

Table 124:   MOH Audio File Formats for Conversion

AUDIO FILE FORMAT AND EXTENSION

Apple (*.aif, *.aifc, *.aiff, *.aiffc)	Psion Record (*.prc)
Amiga 8SVX (*.8svx)	Psion (*.wve)
AU Format Sound (*.au, *.snd)	Sample Vision (*.smp)

Table 124:   MOH Audio File Formats for Conversion (continued)

AUDIO FILE FORMAT AND EXTENSION

Audio Visual Research (*.avr)	Sound Blaster (*.voc)
Compact Disc Digital Audio (*.cdda, *.cdr)	SoundTool (*.sndt)
CVSD Modulation (*.cvs, *.cvsd)	SPHERE (*.nist, *.sph)
IRCAM SDIF (*.ircam, *.sf)	VMS (*.dvms, *.vms)
MAUD File (*.maud)	Wave Sound (*.wav)
Maxis XA (*.xa)	Yamaha TX-16W Sampler (*.txw)

Table 125 lists other file formats that are used by the MiVoice Office 250 MOH Utility. Some of these formats may need additional configuration to use (that is .gsm) or that only a subset of that file type works (that is .m3u, .hcom, .dat).

Table 125:   Other File Formats in the MOH Converter Utility

OTHER FILE FORMATS AND EXTENSIONS

GSM File (*.gsm)	This file requires an external library.
M3U File (*.m3u)	This file is a playlist format, not an audio file format.
Macintosh HCOM (*.hcom)	These files may not convert properly. Convert the files to another common format, such as .aiff or .wav.
Text Data File (*.dat)	This file is a SoX-specific text-file representation of an audio file meant for importing into other analysis tools.

Use the MiVoice Office 250 MOH Utility locally to convert the files to the proper format, and then upload the MOH files to the MiVoice Office 250 using MiVoice Office 250 Administrative Web Portal (AWP). Refer to the AWP Help for more information. After a file is converted to the .n64u format, you cannot run that file format through the converter again. The MOH files are stored on the compact flash-type memory card, but they are not included in the Database Save or Voice Processor Save. When you create your MOH files, make sure you save local copies of both the original music file as well as the converted file.

PROGRAMMING FILE-BASED MOH SOURCES

The following describes how to program File-Based MOH sources.

To program a File-Based MOH source:

1.   Use the MiVoice Office 250 MOH Utility locally to convert the files to the proper format. Refer to the MiVoice Office 250 MOH Utility Help for details.

2.   Upload the MOH files to the MiVoice Office 250 using the Administrative Web Portal (AWP). Refer to AWP Help for more information (System Management/MOH Files). The MOH files are stored on the compact flash-type memory card, but they are not included in the Database Save or Voice Processor Save. When you create your MOH files, make sure you create a backup of your local files.

3.   Use DB Programming to select the audio file to use for MOH (see the applicable audio settings section). You can program a file-based MOH source wherever you are able to

program the existing MOH port. Generally, you can program a file-based MOH source in the following fields in DB Programming (refer to the *MiVoice Office 250 DB Programming Help* for details):

- Audio for Calls Camped onto this Device

- Audio for Calls Ringing this Device

- Audio for Calls Holding for this Device

- Audio on Hold for Transfer Announcement

- Music-On-Hold

- Local Music Source

After a file-based MOH source is created and assigned a filename and when a VoIP resource is available, the audio begins to play immediately and continues until it is unequipped. The VoIP associated with the source is always in use by that MOH source. If the system is oversubscribed and a VoIP resource is not available, the MOH audio file camps on until a VoIP resource becomes available. Any device that attempts to listen to the MOH audio source hears silence. Each file-based MOH source consumes a VoIP resource and a software license, up to 5 audio files. If you unequip a file-based MOH source while a device is playing the file, silence is heard.

The MOH source licensing works differently than IP phone licensing, where an IP phone can obtain a license from a formerly equipped phone and a MOH source cannot. For example, you have 3 filed-based MOH sources licensed and programmed, but then you upload a new license with only 2 MOH licenses. When you upload the new license, the system resets and only the first 2 MOH sources will come online. If you unequip one of the MOH sources with a license, the one without a license will not obtain the newly available license. You must reset the system to reallocate the licenses.

CREATING FILE-BASED MOH PROFILES

"File-Based MOH" is the value available in DB Programming wherever you can change the MOH value. Existing MOH values include Silence, Tick Tone, Ringback, MiVoice Office 250, and Use Next Device's Audio Source (for CO Trunk groups only). When you select "File-Based MOH" for the MOH value, you also must specify the MOH profile as the Extended Value. The Extended Value is the MOH profile that you set to specify which file-based source number to use. The Extended Value is created in the File-Based MOH folder in the System folder.

NOTE	The power of all signal energy other than live voice cannot exceed -9dBm when averaged over a 3 second interval. This means that the File-Based MOH file cannot exceed -12 dBm0 when averaged over a 3 second interval. If any gain on the system (for example, the transmit gain on a loop start trunk) is increased, this maximum level must be decreased by the same amount.

By default, you can create up to five MOH profiles. If there is a MOH profile that is assigned to a file-based MOH source that no longer exists on the system, a warning message appears when you connect to DB Programming.

To create a MOH profile and assign a MOH source to the profile:

1. Select System – **File-Based MOH**.

2. Right-click anywhere in the right pane, and then select **Add to File-Based MOH List**. The Get ID dialog box appears.

3. Select the starting ID and the number of IDs to create. By default you can create up to five entries or the number of File-Based MOH licenses the system has.

4. Click **OK**. The items are added to the list with default values.

5. Click the **Description** column to open an edit box, and then type a description. Click in another area of DB Programming to save your changes.
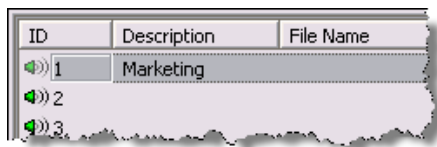
6. Right-click in the File Name column, and then select **Assign File**. The Assign File-Based Music-On-Hold dialog box appears. This dialog box shows the existing MOH files that reside on the compact flash-type memory card.

7. Select the audio file, and then click **Assign**. The MOH profile is configured.

To unassign a file-based MOH source from a profile:

Right-click the File Name, and then select **Unassign**.

To delete a MOH profile:

Right-click the File Name, and then select **Remove Selected Items**.

USING A FILE-BASED MOH SOURCE

The option to choose the file-based MOH appears anywhere in DB Programming where a music source is currently programmable. The File-Based MOH source is included with the following existing music sources: Silence, Tick Tone, Ringback, MiVoice Office 250, and Use Next Device's Audio Source (for CO Trunk Groups only). When you select File-Based MOH for the Value, you must also specify the MOH profile for the Extended Value. **Table 126** shows File-Based MOH options:

Table 126: File-Based MOH Option Fields

FIELD	LOCATION
Audio for Calls Camped onto this Device	• System\Devices and Feature Codes\Nodes\<*node*> • System\Devices and Feature Codes\Hunt Groups\Local\<*hunt group*> • System\Devices and Feature Codes\CO Trunk Groups\<*CO trunk group*> • System\Devices and Feature Codes\Phones\Local\<*phone*> • System\Devices and Feature Codes\Phantom Devices\<*phantom device*> • System\Devices and Feature Codes\SIP Peers\SIP Trunk Groups\<*SIP trunk group*>\Trunk Group Configuration

Table 126: File-Based MOH Option Fields (continued)

FIELD	LOCATION
Audio for Calls Ringing this Device	• System\Devices and Feature Codes\Hunt Groups\Local\<*hunt group*> • System\Devices and Feature Codes\Phantom Devices\<*phantom device*> • System\Trunk-Related Information\Music-On-Hold Profiles
Audio for Calls Holding for this Device	• System\Devices and Feature Codes\Phones\Local\<*phone*> • System\Devices and Feature Codes\Phantom Devices\<*phantom device*> • System\Trunk-Related Information\Music-On-Hold Profiles\<*ID*>
Audio on Transfer to Ring	• System\Devices and Feature Codes\CO Trunk Groups\<*CO trunk group*> • System\Devices and Feature Codes\SIP Peers\SIP Trunk Groups\<*SIP trunk group*>\Trunk Group Configuration
Audio on Transfer to Hold	• System\Devices and Feature Codes\CO Trunk Groups\<*CO trunk group*> • System\Devices and Feature Codes\SIP Peers\SIP Trunk Groups\<*SIP trunk group*>\Trunk Group Configuration
Audio on Hold for Transfer Announcement	• System\Devices and Feature Codes\CO Trunk Groups and Node Trunk Groups • System\Trunk-Related Information\Music-On-Hold Profiles\<*ID*> • System\Devices and Feature Codes\SIP Peers\SIP Trunk Groups\<*SIP trunk group*>\Trunk Group Configuration
Local Music Source	• System\Devices and Feature Codes\Node IP Connection Groups\<*IP connection group*>\IP Call Configuration
Music-On-Hold	• System\Devices and Feature Codes\CO Trunk Groups and Node Trunk Groups. • System\Trunk-Related Information\Music-On-Hold Profiles\<*ID*> • System\Devices and Feature Codes\SIP Peers\SIP Trunk Groups\<*SIP trunk group*>\Trunk Group Configuration

To associate a MOH profile to a file-based MOH source:

1. Click the **Value** column for the specific music source field you are programming, and then select **File-Based MOH**.

2. Right-click the **Extended Value** column, and then select **Change Extended Value**. The Change Extended Value dialog box opens.

3. Select **File-Based MOH**, and then click **Next**.

4. Click the MOH profile, and then click **Finish**. The extended value is populated with the MOH profile number.

TROUBLESHOOTING FILE-BASED MOH

Table 127 lists troubleshooting information for the File-Based MOH feature.

Table 127:   File-Based MOH Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
A subscriber tries to listen to the file-based MOH source. No audio is heard and the phone displays "Waiting for Resources."	The defined file-based MOH source is configured correctly and has an allocated VoIP resource, but the phone does not play the associated audio if there are no VoIP resources available for the phone.	The subscriber hears silence until a VoIP resource becomes available.
A subscriber tries to listen to the file-based MOH source, but no audio is heard.	The file-based MOH source has a filename configured, but a corresponding audio file does not exist. Either the file-based MOH source was deleted or it is not modified to use the proper file in DB Programming.	Reconfigure the file-based MOH source to use the proper music file.
A subscriber tries to listen to the file-based MOH source, but no audio is heard. The file-based MOH source is configured correctly.	The device is using a file-based MOH source that cannot obtain a VoIP resource because there are not enough licenses for the list of defined file-based MOH sources.	The file-based MOH source that does not have a VoIP resource cannot play audio until enough of the other file-based MOH sources are unequipped, therefore freeing up an associated license.
A subscriber tries to listen to the file-based MOH source, but silence is heard. The file-based MOH source is configured correctly. VoIP resource and software license is available.	The file-based MOH source is not in the correct format.	Audio files for this feature must be in the G.711 format. The IP Resource Application (IPRA) still plays back a file even if it is not in the correct format. Convert the file to the proper format before you upload it to the compact flash-type memory card.
A subscriber tries to listen to the file-based MOH source, audio is heard, but the correct audio is not.	The file-name associated with the file-based MOH source is not the correct file.	Associate the correct file name with the desired file-based MOH source in DB Programming.
You cannot upload an audio file to the MiVoice Office 250 using the Administrative Web Portal (AWP).	The compact flash-type memory card is at 80% capacity.	Delete some files on the compact flash-type memory card (either MOH audio files or voice mail messages) to make space available.
You cannot associate a filename with a particular file-based MOH source in DB Programming.	The filename is already in use by another file-based MOH source.	This is as-designed. Use another filename for the file-based MOH source.

PAGE ZONES

Page zones determine the devices that receive system pages. All phones and trunks within a page zone must reside on the same node as the page zone. However, a page zone can contain external *page ports* that are located on other nodes (see **page 750**), if they are programmed as off-node devices on the local node (see **page 367**).

The figure below shows the DB Programming Page Zone location.
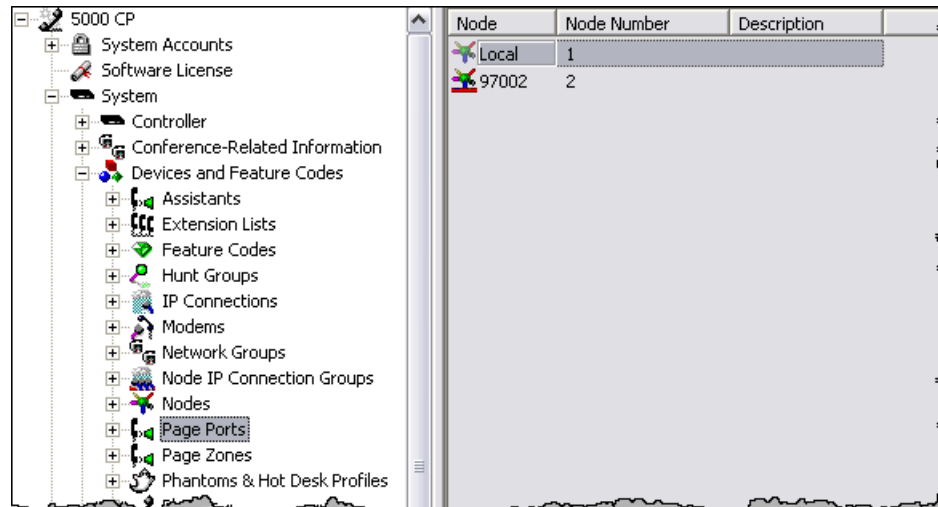
This section contains the following programming instructions:

- **Viewing Page Zones** below
- Planning Page Zones on **page 748**
- Programming Local Page Zones on **page 748**
- Creating Remote Page Zones on **page 749**
- Deleting Page Zones on **page 750**
- Deleting Items from a Page Zone on **page 750**
- Creating Off-Node Page Ports on **page 750**
- Deleting Off-Node Page Ports on **page 750**

VIEWING PAGE ZONES

You can view page zones for local or remote nodes.

To view a page zone:

Select System – Devices and Feature Codes – Page Zones – ***<node>***. Available page zone nodes appear in the right pane.

PLANNING PAGE ZONES

Page zone programming differs for Local Page Zones and Remote Node Page Zone. For more information, see "Programming Local Page Zones" below and "Creating Remote Page Zones" on **page 749**.

NOTICE
Placing a large number of phones in a paging zone may affect system performance. If system operation is affected when a page is placed to a particular page zone, remove some phones from that zone or change to external paging for the area served by that page zone. A significant number of pages between IP phones also increases bandwidth usage and impact system performance.

To prepare for page zone programming:

Make a list of the phones, trunks, and/or the external paging port(s) included in local paging zones. Devices can be in more than one page zone. In the default state, all phones are assigned to page zone 1.

PROGRAMMING LOCAL PAGE ZONES

Along with the page zone, program the phones, trunks, and page ports that are included in the page zone.

CREATING LOCAL PAGE ZONES

To create a local page zone:

1. Select System – Devices and Feature Codes – Page Zones **– <node>**.

2. In the right pane, right-click on the node, and then click **Create Page Zone**. The Create Page Zone Extension dialog box appears.

3. Enter the starting extension number and the number of extensions.

4. Click **OK**. The page zone is added to the list.

5. Double-click the page zone extension number.

6. Enter the following information for the page zone:

    • **Number**: Type the page zone number that system users enter on their phones when placing pages.

    • Description and Username: Descriptions can have up to 20 characters; usernames can have up to 10 characters. Do not use slash (/), backslash (\), vertical slash (|), tilde (~), or Control characters.

7. Add the page zone items. Continue to "Assigning Items to Local Page Zones" on **page 749**.

To change several page zone extension numbers at once:

1. Select the zones you want to change.

2. Right-click, and then select **Batch Extension Change**. The Get Extension screen appears.

3. Select the number you want to assign to the first selected zone (the other selected zones will be numbered consecutively after this number). You can use the SHIFT or CTRL keys to select more than one number.

4. Click **OK**. The page zones are automatically renumbered and re-sorted in the list.

ASSIGNING ITEMS TO LOCAL PAGE ZONES

After creating the page zone, you must add the items that receive the paging messages. You must create Page Ports before you can add them to the Page Zone. See the following section, "Creating Off-Node Page Ports" on **page 750**.

To assign phones, trunks, or paging ports:

1. Select System – Devices and Feature Codes – Page Zones **–** <**node**>.

2. Double-click the page zone that you want to program.

3. Double-click **IP/Digital Telephones**, **Trunks**, or **Page Ports and Off-Node Page Zones** to add the devices to the page zone.

4. Right-click anywhere in the right pane. An option box appears.

5. Select **Add To List**. A window appears prompting for the device type to include.

6. Select the device types (you can use the SHIFT or CTRL key to select more than one item), and then click **Next**. The items with details appear. To view items in a list only, click **List**.

7. Click **Add Items**, and then click **Finish**.

CREATING REMOTE PAGE ZONES

Off-node page zones allow paging access to paging zones located on other nodes. When you view remote node options, a list of its existing off-node page zones appears.

To create Off-Node Page Zones:

1. Select System – Devices and Feature Codes – Page Zones **–** <**node**>.

2. Right-click anywhere in the right pane, and then click **Create Off-Node Page Zone**. The Create Page Zone Off-Node dialog box appears.

3. Enter the starting extension number and the number of extensions.

4. Click **OK**. The page zone is added to the list.

5. Double-click the page zone extension number.

6. Enter the following information for the page zone:

- **Number**: Type the page zone number that system users enter on their phones when placing pages.

- **Description and Username**: Descriptions can have up to 20 characters; usernames can have up to 10 characters. Do not use slash (/), backslash (\), vertical slash (|), tilde (~), or Control characters.

DELETING PAGE ZONES

To delete Off-Node Page Zones:

1. Select the page zone(s).

2. Right-click, and then select **Delete**. The page zone is automatically removed from the list.

DELETING ITEMS FROM A PAGE ZONE

To delete an item from one of the lists:

Select the list item(s), right-click and select **Remove Selected Items**.

CREATING OFF-NODE PAGE PORTS

You must add Off-Node Page Ports for each off-node page zone.

To create an off-node page port:

1. Select System – Devices and Feature Codes – Page Ports – <*remote node*>.

2. Right-click in the right pane, and then click **Create Off-Node Page Port**.

3. Enter the description and user name for the page port. Descriptions can have up to 20 characters; usernames can have up to 10 characters. Do not use slash (/), backslash (\), vertical slash (|), tilde (~), or Control characters.

DELETING OFF-NODE PAGE PORTS

To delete Off-Node Page Ports:

1. Select the page port(s).

2. Right-click and select **Delete**. The page port is automatically removed from the list.

SYSTEM FLAGS

This section describes system-wide flags. Phone flags are described on **page 417**.

To program System Flags:

1. Select System – **Flags** to view a list of all system-wide flags.

2. Select the flag.

3. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box. Some flags require you to choose between two settings (such as 12- or 24-Hour Time Display Format). For those flags, select the current Value and use the scroll box to select the desired value.

4. Click out of the field or press **ENTER** to save your change.

Table 128 shows system flags in the order they appear in DB Programming:

Table 128:   System Flags

FLAG	DESCRIPTION
Agent Help Display	Determines whether phone users will see the Agent Help displays during an Agent Help call. In the default state, this flag is set to Yes.
Agent Help Tone	Determines whether the parties on a call will hear a tone when a supervisor joins or monitors a call using the Agent Help feature. The frequency of the tones are determined by the Agent Help Tone Interval timer, described on **page 763**. In the default state, this is set to Yes.
Allow Green LEDs	*Green LEDs to not apply to multi-protocol phones.* Gives the installer the option of having all red/green phones and red-only phones operate the same, or allowing the red/green phones to use the Green LEDs. If this flag is enabled, the dual-color phones will use both the red and Green LEDs. If it is disabled, the phones will operate in red-only mode, like the phones without Green LEDs. In the default state, this is set to No. See also "Use Green LEDs for Direct Rings," which follows in this list of flag descriptions.
Allow Immediate ACD Auto Connect After DND	Allows calls to autoconnect to the ACD agent phone immediately after the agent removes the phone from DND. If this flag is disabled, the first call received after DND is disabled rings until the agent answers the phone. By default, this flag is disabled
Alternate IP/Digital Telephone Menu Display	Determines the display that Executive Display, Professional Display, and 8560 digital telephone users see during an outside call. In the default state, the main menu display includes the TRANSFER-TO-HOLD option. If alternate displays are enabled, the FLASH option appears in its place. In the default state, this flag is set to No.
Audible Message Indication for SL Sets	If enabled, single line phones will receive message waiting indications (six tones before dial tone) when a message is waiting. In the default state, this is Yes.
Audio Diagnostics Alarm Suppression	Gives you the option of suppressing Alarm 128 (see "Alarms" on **page 1127**), which is generated when the Audio Diagnostics feature (see **page 1113**) is used. By default, this flag is set to No, which means that the alarm is generated and displayed on the administrator's phone when a user access the Audio Diagnostics feature. If this flag is set to Yes, the alarm is suppressed

Table 128:   System Flags (continued)

FLAG	DESCRIPTION
Automatic Board Equip	If this flag is set to Yes, all modules inserted into unequipped bays are automatically detected and the bay is equipped to reflect the inserted module when the system is powered up and a new programming session is started. (All ports on inserted modules are unequipped.) This simplifies initial configuration, because all modules can be installed into the equipment chassis and they will appear in DB Programming automatically, reflecting the actual configuration of the system. In the default state, this flag is set to Yes.
	The Auto Board Detect feature can only function when a DB Programming session is not active.
	The Auto Board Detect function applies only to unequipped bays. Changes of bay type are not automatically detected and previous programming of modules and ports is left intact when module types are changed in previously used bays. Therefore, use local programming mode and delete any modules that exist in bays you want automatically detected before beginning the direct or remote programming session.
Barge-In Notification Display	Controls whether the agent sees the BARGE-IN PROGRESS or BARGE-IN TERMINATED display whenever a supervisor barges-in or hangs up after barging-in. The default setting is Yes. To turn on Barge-In Notification Display, set this flag to Yes. To turn off the Barge-In displays, set this flag to No.
	If the supervisor attempts to steal the other party away from the agent, the steal fails. The other party may be another phone or a trunk on a different node. The supervisor's phone displays VERSION MISMATCH STEAL REJECTED. See "Hunt Group Supervisors and Barge-In" on for details.
Barge-In Notification Tone	When this flag is turned on (set to Yes), each of the barged-in parties (not including the supervisor) can hear a tone on every barged-in call. The tone repeats at the frequency set by the Barge-In Notification Tone Frequency Timer (below). In the default state, this flag is set to No. See "Hunt Group Supervisors and Barge-In" on for details.
Broadcast Alarms To All Administrators	Determine whether system alarms will display on all administrator phones (set to "yes"). If not, they will display only at the primary attendant's phone (set to "no"). In the default state, this is set to Yes.
Broadcast Station Off-Hook Alarms	Determine whether off-hook alarms will be displayed on all administrator phones and the primary attendant's phone (set to "Yes"). In the default state, this is set to Yes.
Companding Type	Determine whether the system uses Mu-Law or A-Law companding.
Date Display Format	Determines the order the Day-Month-Year appear in the system. This includes SMDR Header, Keyset Date Programming, and Message Print. In the default state, it is set to Month-Day-Year.
DISA Transfer Tone	Select the ring or music option. If ring tones are enabled, callers will hear ringing instead of music when using DISA to call a phone. In the default state, this is set to Ring.
Disable Confirmation Tone to Trunks	Normally, a confirmation tone is played on a CO trunk call transferred to a Call Routing Announcement (CRA). This flag disables the confirmation tone for this type of transfer. It is set to No by default.
Disable Mute Confirmation Tone	The mute confirmation tone is played through speaker when a user enables/disables the mute feature. When users on speakerphone use the mute key some of the audio from the confirmation tone was being heard by the far end party. The Disable Mute Confirmation Tone flag disables this tone. It is set to Yes by default.

Table 128:   System Flags (continued)

FLAG	DESCRIPTION
Display Caller ID Name and Number	Adjusts the incoming CO call display to always show caller ID name and number on the LCD display instead of the call timer. For example, when a user is on a call, the user's phone displays a timer that counts the duration of the call. Setting this flag to Yes removes the timer and instead, shows the caller ID (name and number) of the caller. It is set to No by default.
Drop Incomplete Outgoing Calls	Occasionally, a phone user seizes a trunk at the exact time a call is coming in on that trunk and the user connects to the incoming call instead of dial tone. This phenomenon is called "glare." If the Drop Incomplete Outgoing Calls option is turned on (set to Yes), the resulting call on that trunk is dropped, as it would be if the user did not dial a valid number. If the option is turned off (set to No), the call remains connected regardless of the number of digits dialed, if any. In the default state, this flag is set to Yes.
Enable Shutdown On Low Battery	This flag allows the system to initiate a system shutdown If the UPS monitor detects a low battery event. It is set to Yes by default. For details about the UPS Monitoring feature, refer to the UPS Monitoring section in the *MiVoice Office 250 Installation Manual* .
Handsfree Announce On System Forward Transfer	When a phone user transfers a call to the principal phone of a forwarding path, this option determines what the caller will hear. If turned on (set to yes), the caller will place a handsfree call to the principal phone and can announce the transfer (the call does not enter the forwarding path). If the option is disabled, the transferring phone user will hear ringing and can announce the call only if it is answered by the principal phone or one of the forwarding points. In the default state, this flag is set to No.
House Phone Mode	*This flag affects single line House Phones only. Other phone users can dial while on-hook and are not affected by this flag.* The flag determines whether a single line House Phone returns dial tone or does not return dial tone after the called party disconnects. At the System level of programming, all House Phones may be set in either Normal mode or Restricted mode. At the individual Phone level of programming, a House Phone must be programmed to dial specific digits as soon as the handset is taken off-hook. The programmed digits may ring a specific phone or a Hunt Group that rings multiple phones. Normal mode allows the user to enter a feature code or place a call after the automatically called number hangs up. Restricted mode prevents the user from performing any operation other than placing a House Phone call. The System default state is Normal.
Insufficient Bandwidth Alarm	Indicates whether the administrator phone displays a message when alarm 032 is generated. Alarm 032 is generated when the IP network does not have enough bandwidth to support the IP call that is currently connected to the device. If enabled (set to Yes), SYS ALARM #32 X<ext> INSUF BAND, where <ext> is the extension number of the affected device, displays on the administrator phone. If disabled (set to No), the phone does not display any information when the alarm is generated. In the default state, this flag is set to Yes.
Music On Hold For IC Calls	Determines whether a user hears music when placed on hold by another phone or when camped onto a resource. If set to No, IC callers never hear music, although they may hear other tones such as tick tone and ringback. In the default state, this flag is set to Yes.
OHVA Enable	Determines whether the Off-Hook Voice Announce (OHVA) feature will be enabled system-wide. In the default state, this flag is set to Yes. OHVA is not supported on IP phones.

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Table 128: System Flags (continued)

FLAG	DESCRIPTION
Play Pre-Record-A-Call Display	Enables the Voice Processor to play a message before the Record-A-Call message. In the default state, this flag is set to No.
Receive Network Alarms	Determines whether the node will receive and display network-wide alarms sent by other nodes in the network. In the default state, it is set to No.
Record-A-Call Display	Determines whether phone users will see the Record-A-Call displays when the feature is used. In the default state, this flag is set to Yes.
Record-A-Call Tone	Determines whether the parties on a call will hear a tone when the Record-A-Call feature is in use. The timing of the tones are determined by the Record-A-Call Tone Interval timer, described on **page 763**. In the default state, this is set to Yes.
Ring Flash On DSS Lamps	When turned on, this option allows DSS/PKM buttons to indicate a ringing call to a phone by showing the ring flash on the associated button. If turned off, the button will show that the phone is busy (solid lamp). In the default state, this flag is set to Yes.
Send Network Alarms	Determines whether a node will broadcast alarms that occur on that node to the rest of the network. In the default state, this flag is set to No.
Single Idle Time for All Hunt Groups	Invokes an Automatic Call Distribution (ACD) algorithm that sends an incoming call to the agent station with the longest idle time in all the hunt group queues to which that station belongs. This flag only affects hunt groups programmed on the local node, where the flag is set. The members of the local hunt groups can be off node, but are included in the pool for the single idle time for all hunt groups. For agent stations belonging to multiple hunt groups, this feature allows calls to be distributed to other stations having the longest idle time, regardless of the station idle time status in an individual hunt group. The default is set to No.
SLC Ring Zones	In some installations, where more than one single line device is connected in series to an SLM circuit, zoned ringing may be necessary. The system can support devices with 3 Ringer Equivalence (REN) units per circuit. However, the system can ring only eight RENs at once. Therefore, if there are more than eight RENs on any Single Line Module, the system-wide flag must be turned on to allow the single line devices to ring in smaller groups (zones). In the default state this flag is turned off.
SPCL Key Required For Feature Code Entry	Depending on the setting of this flag, users can either enter feature codes immediately after lifting the handset or while on hook, or they must press the SPCL button before entering the feature code. If this flag is turned on, phone users must always press the SPCL button before entering a feature code. In the default state, this flag is turned off (set to No).
System Speed Dial Override Toll Restriction	If this option is turned on, System Speed Dial numbers can be speed-dialed at any phone regardless of toll restrictions. If the flag is turned off, all System Speed Dial numbers are subject to toll restriction. In the default state, this flag is turned off (set to No).
Time Display Format	Determines whether phone displays will show the time in 24-hour or 12-hour (AM/PM) format. In the default state, it is set to 12 hour.
UCD/ACD Station Monitor Indications	Determines whether phone monitoring tones are sent to a UCD or ACD hunt group member when the hunt group supervisor is monitoring the call. In the default state, this flag is turned ON. The frequency of the tone is determined by the UCD/ACD Station Monitor Indication Frequency timer.
	NOTICE: Call monitoring may be illegal in some areas. The end user is ultimately responsible for ensuring that use of this feature is in compliance with applicable laws

Table 128:   System Flags (continued)

FLAG	DESCRIPTION
Use Green LEDs for Direct Rings	Determines whether direct ring-in calls illuminate a red LED lamp or a green one. If the flag is turned on, direct ring-in calls to the phone illuminate a green LED. If turned off, all direct ring-in calls illuminate red. In the default state, this flag is turned on. The Allow Green LEDs flag must also be turned on for this feature to work
Validate Voice Mailbox Numbers	When this flag is enabled, the system checks that a dialed mailbox number matches a programmed extension number when the Record-A-Call or transfer-to-voice mail features are used. In the default state, this flag is set to Yes.
Wrap-Up Mode For Holding ACD Calls	If turned on, this flag prevents an agent from receiving additional ACD hunt group calls while a call the ACD agent placed on hold is holding and the agent's phone is idle. The phone can still receive non-ACD calls, as usual. If the flag is turned off, the agent will be available to receive additional ACD calls as soon as an ACD call is placed on hold. In the default state, this flag is turned off (set to No).

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TIMERS AND LIMITS

You can program the timers and limits that control various system functions. The default values have been carefully selected to ensure proper system operation under most circumstances. Occasionally, you may need to adjust one or more of the timers.

To change the value of a system timer:

1. Select its current Value and then type the new value in the text box.

2. Press **ENTER** or select another field to save the change.

The timer definitions are shown in **Table 129** and on the following pages. A "DID [DDI]" in the timer names indicates that the timer applies to DID [DDI] trunks only. "Digital/IP" refers to phones only and "SL" applies to single line phones only. "E&M" timers apply only to E&M trunks (except E&M Disconnect Flash Duration). "LS" indicates a loop start trunk timer, "GS" refers to ground start trunks, and "LS/GS" applies to both types. The "UCD" timer applies only to UCD hunt groups.

Table 129:   System Timers

TIMER	DEFAULT VALUE U.S. [EUROPE]	RANGE	PURPOSE
Abandoned Call	10	1–255 min	After a call has recalled to the last possible phone, it recalls until this timer expires. If it remains unanswered, the system disconnects the call. This timer is starts when the call begins recalling the first recall phone.
Agent Help Tone Interval	0	0–1000 sec	The system can be programmed to send a tone to the call parties when a supervisor joins or monitors a call using the Agent Help feature. This timer determines how often the tone will be heard. If you set this timer to 0, there will be a single tone when the supervisor joins the call, if tones are enabled, but there will not be periodic tones through out the call.
Background Keyset Update Date and Time	360	60–360 min	Determines how often date and time displays are updated.
Barge-In Notification Tone Frequency Timer	0	0–255	Controls whether the agent sees the BARGE-IN PROGRESS or BARGE-IN TERMINATED display whenever a supervisor barges-in or hangs up after barging-in. See "Hunt Group Supervisors and Barge-In" on **page 568** for details.
BT CLIP Caller ID Wait	5	5–8	Amount of time the system will listen for a CALLER ID (BT CLIP in the UK) message when receiving an incoming call on a Loop Start trunk.

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Table 129:   System Timers (continued)

TIMER	DEFAULT VALUE U.S. [EUROPE]	RANGE	PURPOSE
BT CLIP post CID delay	9	5–10	Amount of time between the end of the Caller ID(BT CLIP in the UK) message and the start of the Ring signal, during an incoming call on a Loop Start trunk.
Camp-On	3	1–255 sec	Amount of time a caller hears busy tone before camping on.
Camp-On Tone	15	5–255 sec	Amount of time between Camp On tones.
CO Reseize	3	1–15 sec	When a user reseizes a trunk, this timer determines the length of time the system will hold the trunk open.
Dial Initiation	15	5–30 sec	Limits the amount of time a phone can remain off hook without dialing before the system sends reorder tones.
DID Disconnect Recognition	150	2–500 ms (.002–.5 sec)	The minimum amount of time the circuit must be on hook for the system to recognize that the call has been disconnected.
DID Off-hook Debounce	10	2–500 msec (.002–.5 sec)	The minimum amount of time the circuit must be off hook before the system will recognize another on-hook/off-hook transition.
DID On-hook Debounce	12	2–500 msec (.002–.5 sec)	The minimum amount of time the circuit must be on hook before the system will recognize another on-hook/off-hook transition.
DID Post-Seize Delay	64	2–500 msec (.002–.5 sec)	The minimum time allowed between the recognition of a seizure and the beginning of digit validation. Used only for Immediate-Dial circuits.
DID Post-Signal Delay	30	2–500 msec (.002–.5 sec)	The minimum time allowed between the end of a handshake and the beginning of digit validation. Used only for Wink-Start and Dial-Delay circuits.
DID Pre-Signal Delay	100	2–500 msec (.002–.5 sec)	The amount of time that must elapse between the recognition of a trunk seizure and the handshake.
DID Ready Timeout	4000	2–65,000 msec (.002–65 sec)	Used only for Wink-Start and Dial-Delay circuits. This is the maximum time the T1 or T1/E1/PRI Module or SLA will wait for the "Digit Register Ready" command. If the timer expires, the SLA or module will automatically initiate a handshake. Used only for Wink-Start and Dial-Delay circuits.
DID Seizure Recognition	30	2–500 msec (.002–.5 sec)	The minimum amount of time the circuit must be on hook for the system to recognize that the trunk has been seized.

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Table 129: System Timers (continued)

TIMER	DEFAULT VALUE U.S. [EUROPE]	RANGE	PURPOSE
DID Signal Hold	200	2–500 msec (.002–.5 sec)	The maximum duration of a wink-start handshake or the minimum duration of a dial-delay handshake. Used only for Wink-Start and Dial-Delay circuits.
Digital/IP Alternate Transient Display Timer	10	1–255 tenths (.1–25.5 sec)	The amount of time transient displays appear on phones with the "Alternate Transient Display Timer" phone flag enabled.
Digital/IP Secondary Extension Key Altering Tone	6	1–1000 sec	This timer is in effect only if the phone has a number greater than 1 in the "Ring When *n* Calls At Extension" field. It determines how often the phone with the secondary extension button will hear an alerting tone while the primary phone has *"n"* number of calls present.
DISA Invalid Extension Failure Limit	3	0–9	Determines how many times a caller will be allowed to enter an incorrect DISA security code without being dropped. To program the failure limit, enter the desired number in the text box. Entering 0 allows unlimited attempts.
DISA Security Code Failure Limit	3	0–9	Determines how many times a DISA caller will be allowed to dial an invalid extension number before being sent to the primary attendant. To program the limit, enter the desired number in the text box. Entering 0 allows unlimited attempts.
Disconnect Wait After Dialing	20	1–60 sec	Length of time the system waits after dialing an outside number before checking the trunk for disconnect.
Door Relay Duration	5	1–30 sec	This timer indicates how long the door relay will be activated when the Activate Door Relay feature is used. See "Door Relay Control" on **page 293** for details about Door Relay Control.
DTMF Digit Duration/ Pause	60	30–255 msec (.03–.255 sec)	Adjusts the duration of and pause between digits of DTMF tones sent by the system. Both the tone and the pause will use the assigned duration (for example, a 6/100 second tone has a 6/100 second pause between digits). **Note:** If using private IP networking, set this value to 100 msec or greater. If this value is less than 100 msec, the DTMF tones may not be detected.
E&M Answer Recognition	4500	2–20,000 msec (.002–20 sec)	(U.S. Only) Determines the minimum amount of time the receiving PBX must be off-hook, when a phone user places an outgoing call on an E&M trunk. This allows the system to recognize that the call has been answered.

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Table 129:   System Timers (continued)

TIMER	DEFAULT VALUE U.S. [EUROPE]	RANGE	PURPOSE
E&M Dial Delay	70	2–500 msec (.002–.5 sec)	(U.S. Only) The maximum amount of time the E&M circuit will wait before transmitting digits following a handshake. This timer goes in effect after the handshake on a Wink-Start or Dial-Delay circuit or after seizure on an Immediate-Dial circuit.
E&M Dial Delay Hold	140	2–500 msec (.002–.5 sec)	(U.S. Only) The minimum length for a Dial-Delay handshake. Used only for Dial-Delay circuits.
E&M Dialing Wait After Hookflash	3000	2-20,000 millisec. (.002–20 sec.)	(U.S. Only) Determines how long the E&M circuit will wait when transmitting a hookflash (recall) before dialing additional digits or checking for disconnection.
E&M Disconnect Flash Duration	15,000	2–40,000 msec (.002–40 sec)	(U.S. Only) The minimum amount of time a T1 E&M or DID circuit remains on-hook to cause a disconnection from the remote circuit.
E&M Disconnect Recognition	1500	2–10,000 msec (.002–10 sec)	(U.S. Only) The amount of time a circuit must be on-hook before the E&M circuit recognizes a disconnection.
E&M False Signal Debounce	50	2–500 msec (.002–.5 sec)	(U.S. Only) Determines the minimum length of a valid handshake signal received from a remote circuit. Used only on Wink-Start and Dial-Delay circuits.
E&M Handshake Timeout	5000	2–20,000 msec (.002–20 sec)	(U.S. Only) Determines the maximum length of a valid handshake signal. Used only on Wink-Start and Dial-Delay circuits.
E&M Hookflash Duration	600	2–10,000 msec (.002–10 sec)	(U.S. Only) Determines the length of hookflashes [recalls] sent to the remote circuits.
E&M Hookflash Recognition	300	2–10,000 msec (.002–10 sec)	(U.S. Only) Determines the minimum length of recognizable hookflashes [recalls] from the remote circuit.
E&M Off-Hook Debounce	10	2–500 msec (.002–.5 sec)	(U.S. Only) The minimum amount of time the remote circuit must be off hook before the E&M circuit recognizes another on-hook/off-hook transition.
E&M On-Hook Debounce	10	2–500 msec (.002–.5 sec)	(U.S. Only) The minimum amount of time the remote circuit must be on hook before the E&M circuit will recognize another off-hook/on-hook transition.
E&M Post-Seize Delay	65	2–500 msec (.002–.5 sec)	(U.S. Only) The minimum allowed time between the recognition of a seizure and the beginning of digit validation. Used only for Immediate-Dial circuits.

Table 129: System Timers (continued)

TIMER	DEFAULT VALUE U.S. [EUROPE]	RANGE	PURPOSE
E&M Post-Signal Delay	30	2–500 msec (.002–.5 sec)	(U.S. Only) The minimum allowed time between the end of a handshake and the beginning of digit validation. Used only for Wink-Start and Dial-Delay circuits.
E&M Ready Timeout	4000	2–20,000 msec (.002–20 sec)	(U.S. Only) **On a Wink-Start circuit:** The maximum time the E&M circuit waits for a "Digit Register Ready" command before initiating the Wink-Start handshake. **On a Dial-Delay circuit:** The maximum time the E&M circuit waits for a "Digit Register Ready" command before the Dial-Delay signal is terminated.
E&M Receive Handshake Delay	20	2–500 msec (.002–5 sec)	(U.S. Only) The minimum time required, after seizure, before an incoming handshake signal can be recognized. Used only for Wink-Start and Dial-Delay circuits. If a Wink or Dial Delay signal is detected within this time, the outgoing call is blocked.
E&M Seizure Debounce	2	2–500 msec (.002–.5 sec)	(U.S. Only) Determines the minimum amount of time the remote circuit must remain off hook before the E&M circuit validates the incoming call.
E&M Transmit Handshake Delay	100	2–500 msec (.002–.5 sec)	(U.S. Only) The minimum delay time between the recognition of a seizure and the beginning of a handshake signal. Used only for Wink-Start and Dial-Delay circuits.
E&M Wait for Dial Tone	1000	2–20,000 msec (.002–20 sec)	(U.S. Only) The amount of time the E&M circuit waits for dial tone before dialing digits on an outgoing call. Used only on Immediate-Dial circuits with the Dial Tone Wait option enabled.
E&M Wink Hold	214	2–500 msec (.002–.5 sec)	(U.S. Only) Determines the duration of a Wink-Start handshake.
E&M Wink Timeout	350	2–500 msec (.002–.5 sec)	(U.S. Only) Determines the maximum allowed duration of wink signals that are received from the remote circuit. If the time limit is exceeded, the call is blocked and the attempt terminated. Used only for Wink-Start circuits.
Forward No Answer	15	3–255 sec	Amount of time a call waits at an unavailable phone before being forwarded. Applies to manual call forwarding only, *not* system forwarding.
GS Dialing Wait After Connect	30	1–50 tenths (.1–5 sec)	(U.S. Only) The amount of time the system waits after a ground start trunk has been seized, to place an outgoing call, before dialing digits. This timer is not used if the Loop Current Dialtone Detection option is selected.

Table 129:   System Timers (continued)

TIMER	DEFAULT VALUE U.S. [EUROPE]	RANGE	PURPOSE
GS Tip-Ground Debounce	50	10–500 msec (.001–.5 sec)	(U.S. Only) The amount of time the system ignores subsequent attempts to seize a ground start trunk once it has been seized. The database contains a phone option that can be set to prevent users from reseizing a trunk. If selected, the phone user cannot reseize a trunk until it is disconnected by replacing the handset, pressing the SPKR button (if off-hook), or pressing another trunk button.
GS Transition Delay	10	10–500 msec (.001–.5 sec)	(U.S. Only) The amount of time that must elapse after a disconnection before another call can ring in on that trunk.
Hold	60	0–255 sec	Limits the time a call remains on hold before recalling the phone. If set to 0, the call will *not* recall.
Hold – Alternate	180	0–1000 sec	If the phone has the Alternate Hold Timer flag enabled, this limits time a call remains on hold before recalling the phone. If set to 0, the call will *not* recall.
Inactivity Alarm	60	10–255 sec	Limits the time a phone can remain off hook and inactive (after first receiving reorder tones) before registering a system alarm.
Interdigit – Long Interdigit – Short	15 4	2–255 sec 2–30 sec	Used in determining end of dialing. Short timer is used after a valid number has been dialed. Long timer is used until digits form a valid number.
Loopback Timeout	600	10–10,000 sec	The maximum allowed duration of a remote loopback test. If the timer expires, the test is automatically terminated.
LS Dialing Wait After Connect	15	1–50 tenths (.1–5.0 sec)	The amount of time the system waits for outside dial tone before dialing or checking the trunk for a disconnect.
LS/GS Caller ID Relay Hold	3747	2–4000 msec. (.002–4.0 sec)	The amount of time the system looks for Caller ID [CLID] information when a call is received.
LS/GS Caller ID Ring Idle	128	128–1920 msec (.128–1.1920 sec)	This sets the time between the end of first ring and the time at which the system begins to check for Caller ID [CLID] information. The Caller ID [CLID] timer values combined must be shorter than the period of silence between rings from the CO [local branch].
LS/GS CO Hookflash	60	2–1000 hundredths (.02–10.0 sec)	Adjusts the duration of the timed hookflash (recall) that is sent over the trunk by the system when the Hookflash [Recall] feature code is used.

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Table 129: System Timers (continued)

TIMER	DEFAULT VALUE U.S. [EUROPE]	RANGE	PURPOSE
LS/GS CO-CO Disconnect	35	2–1000 hundredths (.02–10.0 sec)	A call is disconnected by the system if it detects loss of loop current lasting longer than this timer during trunk-to-trunk calls.
LS/GS Dialing Disconnect	120	2–1000 hundredths (.02–10.0 sec)	The connection is dropped if the system detects loss of loop current lasting longer than this timer during dialing.
LS/GS Dialing Wait After Hookflash	30	2–199 tenths (.2–19.9 sec)	Delays dialing after a hookflash [recall] to allow the system and central office hardware to recover.
LS/GS IC-CO Disconnect	60	2–1000 hundredths (.02–10.0 sec.)	During phone-to-trunk calls, the system disconnects a call when it detects loss of loop current lasting longer than this timer setting.
LS/GS Inter-ring Silence	60	1–250 tenths (.1–25.0 sec.)	Indicates the duration of the silence between rings on an incoming call to determine if the trunk has stopped ringing prior to being seized. In most areas, trunk ring pattern is 2 seconds on/ 4 seconds off. Check with the local service provider for the ring pattern in your area. **Note:** This timer must always be set higher thanthe central offices ring off time.
LS/GS Loop Current Debounce	100	2–500 msec (.002–.5 sec)	The minimum amount of time the system must detect loop current for it to recognize that a trunk is present when it is seized.
LS/GS Ring Frequency – High Boundary	100	1000 Hz maximum	These parameters determine the valid range of ring frequencies that will be recognized by the system.
LS/GS Ring Frequency – Low Boundary	15	4 Hz minimum	Any ring signal outside of this range is ignored. The ranges for the Ring Frequency timers are interdependent. The minimum value for the high boundary is the current Value of the low boundary. The maximum for the low boundary is the current Value for the high boundary.
LS/GS Trunk Ring Detection	150 [150]	100–1024 msec (.1–2.5 sec)	A low-level timer that specifies the duration that continuous ring voltage must be detected on a trunk for the system to recognize a new incoming call. If this timer is too low, then false rings could be detected. If the timer is too high, then new incoming calls may not be detected at all. Common ring durations sent by central offices [local exchanges] are 1-second and 2-second.
Message Wait	5	1–255 sec	Amount of time a caller waits after pressing the MSG button before being connected to the called party's message center.

Table 129:   System Timers (continued)

TIMER	DEFAULT VALUE U.S. [EUROPE]	RANGE	PURPOSE
Off-Hook Voice Announce Screening	5	1–255 sec	After the Camp On timer expires, length of time before an OHVA call can be completed. OHVA is *not* functional on IP phones.
Page	15	0–255 sec	Limits duration of page. If set to 0, pages are unlimited in length.
Pause Dialing Digit Length	3	1–15 sec	Duration of timed pauses used in System and Station Speed Dial numbers and in ARS dial rules.
Queue Callback	15	10–255 sec	Time allowed for a phone to respond to a queue callback before the queue is canceled.
Recall	60	10–255 sec	Amount of time a Hold or Transfer recall rings at a phone before recalling that phone's recall destination. If the phone receiving the recall has no recall destination, the call remains at the phone until the Abandoned Call timer expires.
Record-A-Call Tone Interval	0	0–255 seconds	The system can be programmed to send periodic tones when the Record-A-Call feature is used. This timer determines how often the tone will be sent. If you set this timer to 0, there will be a single tone when the Record-A-Call feature begins, if tones are enabled, but there will not be periodic tones through out the call.
Remote Programming Invalid Extension Failure Limit	3	0–9 failures	Determines how many times a caller will be allowed to enter an incorrect extension number while attempting remote programming, without being dropped. To program the failure limit, enter the desired number in the text box. Entering 0 will allow unlimited attempts.
Remote Programming Passcode Failed Limit	3	0–9 failures	Determines how many times a caller is allowed to dial an invalid extension passcode while attempting Remote Programming, before being disconnected. To program the limit, enter the desired number in the text box. Entering 0 allows unlimited attempts.
Routing Step Advance Timer	8	1–1000 seconds	Serves as the system-wide default for the Dynamic Extension Express Steps. When a Dynamic Extension Express step is created, this field is used to default the Step Advance Timer field for the Step. It signifies the amount of time that each routing step rings before advancing to the next routing step.

Table 129:   System Timers (continued)

TIMER	DEFAULT VALUE U.S. [EUROPE]	RANGE	PURPOSE
SL Disconnect Flash Duration	15	1–250 tenths (.1–25.0 sec)	When a single line phone is involved in a call that is dropped by the other party, and the single line phone remains off-hook, the system turns off tip and ring battery for the duration of this timer. If peripheral equipment is connected to the single line circuit (such as a voice mail unit, page amplifier, or other system) then this loss of battery signals a call disconnect and usually causes the equipment to disconnect.
SL Hookflash Maximum	12 [15]	2–20 tenths (.2–2.0 sec)	The maximum amount of time a single line phone user can press the hookswitch before the system disconnects calls.
SL Hookflash Minimum	2 [7]	1–10 tenths (.1–1.0 sec)	The minimum length of time a single line phone user must press the hookswitch for the system to recognize a hookflash [recall].
SL Wait For Disconnect	2	1–60 seconds	When a single line phone is involved in a call that is disconnected by the other party, and the single line phone remains off hook, this timer specifies the length of time between the disconnection and when the disconnect flash (which is set by the Disconnect Flash Duration timer) is transmitted to the single line circuit. During this time, the phone receives no audible signal.
System Forward Advance	15	2–255 seconds	Determines how long a call will ring (unanswered) at each forwarding point in a system forwarding path.
System Forward Initiate	15	2–255 seconds	Determines how long a call will ring (unanswered) at the principal phone before advancing to the first forwarding point in a system forwarding path.
Transfer – Attendant	30	10–255 seconds	When an attendant transfers a call, this timer limits the time a transferred call rings unanswered before it recalls the attendant.
Transfer – Available	20	10–255 seconds	Limits time a transferred call rings unanswered before it recalls the transferring phone. (Does **not** apply to calls transferred by attendant phones or the Voice Processor or calls transferred to a busy phone.)
Transfer – Busy	24	10–255 seconds	Limits time a transferred call waits at a busy phone before recalling the transferring phone. Does **not** apply to calls transferred by attendant phones or Voice Processor or to calls transferred to an idle phone.

Table 129: System Timers (continued)

TIMER	DEFAULT VALUE U.S. [EUROPE]	RANGE	PURPOSE
Transfer – Voice Processor	20	10–255 seconds	Limits time a transferred call waits at a Voice Processor destination before recalling the programmed recall destination.
Trunk Key Debounce	3	1–30 seconds	The amount of time the system ignores subsequent attempts to press a trunk button once it has been pressed. This timer prevents a user from accidentally disconnecting a call by reselecting the trunk button if the button is pressed twice while answering a call.
			The database contains a CO/IC Reseize phone flag that can be set to prevent users from reseizing a trunk. If selected, the phone user cannot reseize a trunk until it is disconnected by replacing the handset, pressing the SPKR button, or pressing another trunk button.
			Note: The database contains a CO/IC Reseize phone flag that can be set to prevent users from reseizing a trunk. If selected, the phone user cannot reseize a trunk until it is disconnected by replacing the handset, pressing the SPKR button, or pressing another trunk button.
UCD/ACD Station-Monitor Indication Frequency	15	1–255 seconds	The duration of the pause between hunt group phone monitoring tones.
Unsupervised CO	5	1–255 minutes	Limits duration of outside calls transferred or forwarded to outside numbers before recalling the primary attendant.
Valid Call	15	1–60 seconds	Minimum duration of an outgoing call before it is recorded in SMDR. Calls placed on hold or transferred are not subject to this timer.
Voice Mail Dial Delay	5	1–250 tenths (.1–2.5 sec)	(Used for Analog Voice Mail Hunt Groups Only) When a call is answered by the voice mail unit, this indicates the amount of time the system waits before sending digits to the voice mail unit after it answers the call.

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FEATURE CODES

System feature codes are preset to carefully selected default values. Changing the codes can erase existing assignments. For example, if 300, 305, and 306 are assigned as feature codes and you attempt to assign 30 as another feature code, you would receive a warning message, because 30 makes up part of existing codes. The warning message allows you to change the existing numbers (300, 305, and 306) individually or to leave the existing numbers unchanged by selecting Cancel.

Feature code tables begin on **page 768**.

Remember the following when changing feature codes:

- If you change feature codes, they are not automatically updated in the feature code directory used by Desktop Interfaces (system devices. You must manually update the directory. However, any devices installed after the change will have the new feature code information.

- Desktop Interface functionality requires the Desktop Interface software license.

To change a feature code number:

1. Select System – Devices and Feature Codes – **Feature Codes**.

2. Select the current Feature Code.

3. Select or enter the new code in the box. Feature code descriptions cannot be changed.

To change several feature code numbers at the same time:

1. Select System – Devices and Feature Codes – **Feature Codes**.

2. Select the codes you want to change.

3. Right-click and select **Batch Extension Change**.

4. When the Get Extension screen appears, select the number you want to assign to the first selected feature code. The other selected codes will be numbered consecutively after this number.

5. Click **OK**. The feature codes are automatically renumbered and resorted in the list. To select a series of items, hold down SHIFT while selecting the first and last item in the range. To select two or more items that are not consecutive, hold down CTRL while selecting the desired items. You may need to continue to hold SHIFT or CTRL while right-clicking to display the option without changing the selected items.

CONFIGURABLE FEATURE/TRUNK LABELS FOR SELF-LABELING PHONES

The Mitel 5320, 5330, 5340, and 5360 IP phones have self-labeling buttons that can be programmed with custom feature and trunk labels, as described below.

FEATURE LABELS

This feature allows system administrators to assign system-wide custom feature labels that will override the default feature labels that normally appear on phones with self-labeling buttons (keys). For example, a customer might prefer "Record" over "Rec-A-Call" for the Record-a-Call Feature.

Note that custom feature labels are language independent. If a custom label is defined in the database, it will be displayed on self-labeling phones regardless of the language chosen for those phones. The system's default feature labels are language specific, so if no custom label is programmed, the label displayed will correspond to the chosen language.
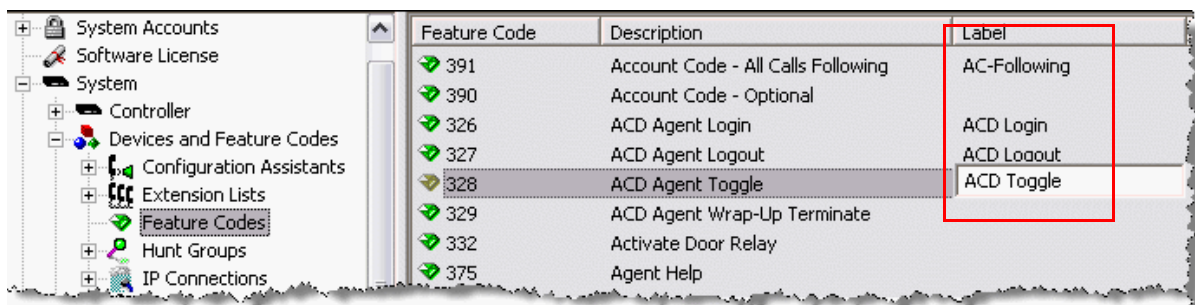
The self-labeling region on self-labeling phones is the only area where the custom labels are used. There are no changes to any standard displays, including the Program Buttons, Review Buttons, and Feature Directory displays. These functions all continue to show the standard default feature names.

TRUNK LABELS

Prior to v4.0, self-labeling phone buttons programmed as trunks displayed the associated trunk group description (that is, trunk group number). Version 4.0 or later system includes a Trunk Label field so that system administrators can assign useful labels (for example, phone numbers or trunk names) for self-labeling phone buttons. If no label is assigned, the button displays the trunk group description. Also note that the assigned label is overridden with Caller ID information when available on an active call and the phone is in small font mode.

The Review Buttons (396) and Program Buttons (397) feature codes will display the trunk label if assigned.

When the label for a feature or trunk is blank (default), the IP phone display appears as it has in the past. When a label is programed, the label appears on the phone, overriding the normal display. The label column appears between the extension and the full name of each feature.

To program the label for a feature or trunk:

1. Select System – Devices and Feature Codes, and then select either **Feature Codes** or **Trunks**.

2. Select the feature or trunk that you want to label.

3. Type the label up to 16 characters in the Label column.

TRUNK ACCESS CODES

Table 130 lists trunk access codes used to select trunks when placing outgoing calls.

Table 130:   Trunk Access Codes

FEATURE NAME	CODE U.S. (EUROPE/ AUSTRALIA)	DEFINITION
Automatic Route Selection (ARS)	92000 (8200 for Australia)	Allows the system to select the route wanted for placing a call, as programmed in the database.
Trunk Group Access 1–208	92001–92208 (82001 for Australia	Selects an available trunk from a programmed group of trunks for placing an outside call.
Emergency Call	911 (999/112/or as applicable)	Entering this feature code selects an outgoing trunk and automatically dials the programmed Emergency Call number, which is routed by default out Trunk Group 1.
Outgoing Call	8 (0 for Australia)	Selects an outgoing trunk according to the programmed outgoing access mode for that phone.

PHONE FEATURE CODES

Table 131 lists the default phone system feature codes.

Table 131: Phone Feature Codes

FEATURE NAME	CODE	DEFINITION
Account Code – All Calls Following	391	Allows the phone user to enter a forced or optional account code that will apply to all calls following the entry of this feature code and will appear in the SMDR. To disable the All Calls Following feature, the feature code is entered again without an account code.
Account Code – Optional	390	Allows the phone user to enter an optional account code for SMDR reports during an outside call.
ACD Agent Login ACD Agent Logout ACD Agent Login/Logout Toggle	326 327 328	These feature codes allow an ACD hunt group member (agent) to log into and out of the ACD hunt group(s). The agent will only receive calls through the ACD hunt group(s) while logged in.
ACD Agent Wrap-Up Terminate	329	When an ACD agent completes a call, no other ACD hunt group call will ring at the phone until the ACD Wrap-Up Timer expires or the agent enters this feature code to terminate the wrap-up session.
Activate Door Relay	332	If the Active Door Relay flag in the System\Devices and Feature Codes\Phones\<*phone*>\Flags folder is enabled, a user can enter the Open Door Relay feature code to activate the door relay. See "Door Relay Control" on **page 293** for details about Door Relay Control.
Agent Help	375	The Agent Help feature allows a phone user to request help from a designated "Agent Help Extension" during a two- or three-party call.
Agent Help Reject	376	When a request-for-help call rings, the Agent Help Extension can choose to join the call or enter this feature code to reject the request.
Answer (Ringing Call)	351	Answers the call that has been ringing or holding the longest at that phone.
Audio Diagnostics	320	When initiated, users are prompted to answer questions about the audio problems by pressing specific keypad buttons.
Automatic CO Access On/Off	360	*(Not used on single line phones)* Allows the phone user to determine how ringing outside calls will be answered: simply by lifting the handset or pressing the Speaker button (automatic answer), or by lifting the handset or pressing the Speaker button and pressing a Call button, individual trunk button or the ANSWER button.
Automatic IC Access On/Off	361	*(Not used on single line phones)* Allows the phone user to determine how ringing intercom calls will be answered: simply by lifting the handset (automatic answer), or by lifting the handset and pressing the IC button (or a Call button, if there is no IC button).

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Table 131:   Phone Feature Codes (continued)

FEATURE NAME	CODE	DEFINITION
Automatic Trunk Answer	350	Using this feature code, phone users with allowed answer can pick up trunks that are ringing into the system, but that are not actually ringing at their phones. This feature does not pick up transferred calls or recalls that are ringing at the phone.
Background Music On/Off	313	*(Not used on single line phones)* Turns on and off background music heard through the phone speaker.
Barge-In	386	Allows the supervisor to barge-in on a call to help the hunt group member/agent.
Call Forward All Calls	355	Immediately forwards all calls to another phone or to an outside phone number.
Call Forward If Busy	357	Immediately forwards all calls to another phone or to an outside phone number when the phone is in use.
Call Forward If No Answer	356	Forwards all calls to another phone or to an outside phone number if not answered within a predetermined time.
Call Forward If No Answer/Busy	358	Forwards all calls to another phone or to an outside phone number if not answered within a predetermined amount of time, or immediately if the phone is in use.
Call Logging	333	Allows users of display phones to view missed, received, and dialed calls.
Change Language	301	A phone user can change the assigned language for the phone by entering the Change Language feature code while the phone is idle.
CO Hookflash	330	Sends a timed hookflash over the trunk while on an outside call (includes conference calls).
Conference	5 or 55 (as applicable)	Connects from three to four parties in a conference. A conference consists of any combination of inside and outside parties.
Data	340	Allows operation of a data device attached to a digital telephone. Requires a modem-equipped data device. Supported on MiVoice Office 250s equipped with appropriate digital interface equipment.
Default Station	394	This single feature code cancels account codes for all calls following, Do-Not-Disturb, manual call forwarding, background music, ring intercom always, and queue requests; restores handsfree mode, pages, hunt group calls, and system forwarding; and returns phone volumes to default values.
Directory	307	*(Display phones Only)* Allows display phone users to search for extension numbers or System Speed Dial numbers. The number can then be dialed, if appropriate.

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Table 131:   Phone Feature Codes (continued)

FEATURE NAME	CODE	DEFINITION
Display Outside Party Name On/Off	379	*(Display phones Only)* When the phone user enters this feature code, while connected to a CO call that has outside party name information, the display will toggle between the caller's name and number. If there is no outside party name or the Expanded CO Call Information On Displays flag is disabled, the user will hear a burst of reorder tone and see the CANNOT ACCESS FEATURE display. If the Display Outside Party Name On/Off feature code is programmed in a user programmable button with a lamp, the lamp will be lit when the outside party name is enabled and off when the outside party number is enabled.
Display Time/Date (ITP) Show IP (SIP)	300	*(Display phones Only)* Temporarily displays the system date and time, user name, and extension number during a call or when other displays are shown. Feature code 300 displays the IP Address of a phone if it is in SIP mode.
Do-Not-Disturb Do-Not-Disturb Cancel Do-Not-Disturb On/Off	370 371 372	The Do-Not-Disturb feature code halts all intercom calls, transferred calls, and pages to the phone. The Cancel code returns the phone to normal operation. The on/off code can be used to turn Do-Not-Disturb on or off.
Do-Not-Disturb Override	373	*(Not used on single line phones)* If enabled in the database, allows the phone user to break through another phone's Do-Not-Disturb mode when placing an intercom call.
Dynamic Extension Express On Dynamic Extension Express Off Dynamic Extension Express On/Off	363 362 364	These feature codes allow the phone user to enable or disable the Dynamic Extension Express feature (a licensed feature).
Dynamic Extension Express Handoff	388	*(Not available if the user's main extension is a SIP extension.)* Allows the following functions: • **Handoff Pull**: Allows the user who answers a call routed to his or her mobile phone to hand off the call back to his or her extension. • **Handoff Push**: Allows the user to transparently send a connected call on his or her extension to the user's mobile phone or other user destination without disconnecting audio. In order to push a call from an internal extension that is not the main extension, the phone needs the Associated User Extension field configured to be the main extension of the User. The Associated User Extension field can be found under the \System\Devices and Feature Codes\Phones\<*extension*>\Associated Extensions. See "Associated User Extension" on **page 394** for details.

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Table 131:   Phone Feature Codes (continued)

FEATURE NAME	CODE	DEFINITION
Emergency Call	1 and 911 for the U.S;. 999 for Europe; 112 for Australia	Entering this feature code selects an outgoing trunk and automatically dials the programmed emergency number
Enhanced Speakerphone Enable	310	*(Digital telephones only)* When entered at a digital telephone, this feature code enables the enhanced speakerphone. Digital telephones can also use the Special button + Speaker buttons.
Feature Key Default	395	*(Not used on single line phones)* Phones have user-programmable feature buttons that can be set to enter feature codes. This code returns the user-programmable buttons to the database default values.
Group Listen	312	*(Not used on single line phones)* Allows a user to transmit a conversation over the phone speaker while in handset or headset mode.
Handsfree On/Off	319	*(Not used on single line phones)* Disables/enables the phone's handsfree intercom answering. Incoming intercom calls ring as private calls if handsfree answering is disabled.
Headset Enable Headset Disable Headset On/Off	315 316 317	*(Not used on single line phones)* The enable code signals the system that a headset has been connected to the phone. The disable code returns the phone to normal operation. The on/off feature code can be used to toggle the feature on or off.
Hold – Individual	336	Places a call on hold so that it can be picked up directly at that phone or through a reverse transfer from any other phone.
Hold – System	335	Places an outside call on system hold. It can be picked up directly at any phone that has an individual trunk button and has allowed-answer and/or outgoing access for that trunk, or by the phone that placed it on hold. (If used on conference or intercom calls, the system places the call on individual hold.)
Hot Desk On/Off	348	Allows the phone user to log on to a Hot Desk-enabled phone or log off of a Hot Desk session. For details about Hot Desking, see "Hot Desk Profiles for Hot Desking" on **page 447**.
Hunt Group Remove Hunt Group Replace Hunt Group Remove/Replace	322 323 324	Removes the phone from its assigned hunt group(s) or places it in again. Does not affect non-hunt group calls. The remove/replace feature code can be used to toggle the feature.
LCD Contrast Adjustment	303	Adjusts the LCD contrast on the display. The phone must be idle to use this feature.
Message	365	This feature code is used for leaving and retrieving a message waiting indication at a called phone or the called phone's message center. Depending on how the message was left, the called phone user either retrieves the message from his/her message center or from the phone that left the message.

Table 131: Phone Feature Codes (continued)

FEATURE NAME	CODE	DEFINITION
Message – Cancel	366	Allows the phone user to cancel a message waiting indication that he or she left at another phone.
Message – Cancel Current	368	Cancels a message waiting indication that is waiting at the phone without requiring the user to respond to it. (Or, press the asterisk [*] button while viewing the message.)
Message – Silent	367	Leaves a Message Waiting indication at a phone without first placing an intercom call.
Mute On/Off	314	*(Not used on single line phones)* Turns the microphone on or off during a call. If muted, the phone user can hear the other party, but the party cannot hear the phone user.
Page	7 or 57 (as applicable)	When followed by a paging zone code (0–9 or 0–49), it allows announcements to be made through phone speakers and any external paging speakers in the page zone.
Page On/Off	325	*(Not used on single line phones)* Halts pages through the phone speaker or allows them to be received again.
Program Bit Rate	393	*Reserved for controlled introduction.* This allows a digital telephone user to change the bit per second (bps) rate of an attached PCDPM serial port.
Program Buttons	397	*(Not used on single line phones)* User-programmable feature buttons and Station Speed Dial buttons can be programmed using this feature code.
Program Passcode	392	The phone passcode is used for the Remote Programming feature. The passcode can be changed by entering the Program Passcode feature code at the phone or when using the Remote Programming feature.
Queue Request	6	Requests (or cancels) an automatic callback when a busy trunk or phone becomes available.
Record-A-Call	385	If the system is programmed with a Record-A-Call application, the phones can be programmed to use the Record-A-Call feature. It allows users to enter a feature code whenever they want to record an ongoing call in their designated Record-A-Call mailbox. Users can retrieve the recorded messages later, just as they would any other mailbox messages.
Redial	380	Redials the last outside phone number dialed or saved at the phone (up to 48 digits). Also used to save numbers at phones programmed for last number saved. (86xx phones use the REDIAL button.)
Redirect Call	331	Allows the phone user to route ringing outside, intercom, and camped on calls to another phone, hunt group, or outside number. Routing of the redirected call is still subject to trunk and toll restrictions. This feature provides these options in addition to the currently available options which allow the phone user to redirect calls to Voice Mail or Do-Not-Disturb. The Redirect Ringing Call feature does not require a software license.

Table 131:   Phone Feature Codes (continued)

FEATURE NAME	CODE	DEFINITION
Reminder Message Reminder Message Cancel	305 306	*(Not used on single line phones)* The phone user can set reminder messages that signal the phone at specific times. Or, the user can cancel all reminder messages for the phone.
Remote Configuration – Disable	343	*Reserved for controlled introduction. Disables the Remote Configuration feature. The VPN connection from the Remote Proxy Server to the MiVoice Office 250 is terminated.*
Remote Configuration – Display HW Serial Number	347	*Reserved for controlled introduction. Displays the hardware serial number for the MiVoice Office 250.*
Remote Configuration – Enable	342	*Reserved for controlled introduction. Enables a Remote Configuration session. The Virtual Private Network (VPN) connection from the MiVoice Office 250 system to the Remote Proxy Server is initiated.*
Remote Configuration – Reset	344	*Reserved for controlled introduction. Resets a Remote Configuration session. The VPN connection from the MiVoice Office 250 system to the Remote Proxy Server is reset.*
Remote Programming	359	Allows a user to place a phone in DND mode, forward the phone's calls, or change the password; either from another phone or through DISA.
Reverse Transfer (Call Pick-Up)	4	Picks up a call ringing or holding at a phone or hunt group. See "Transfer – Reverse Transfer" on **page 339** for details.
Review Keys	396	*(Not used on single line phones)* User-programmable feature buttons and Station Speed Dial buttons can be viewed using this feature code.
Ring Intercom Always On/Off	377	Enables/disables the feature that allows the phone to always place private (non-handsfree) intercom calls.
Ring Tone Selection	398	*(Not used on single line phones)* Selects the type of ringing alert tone that will be heard from the phone.
Routing Off	304	Disables System OAI Offering Control for third-party applications. This feature requires the user to enter a passcode. Once you disable routing, you cannot enable it again (that is, only the third-party application can enable routing).
Station Monitor	321	*(Hunt Group supervisors only)* Allows a designated hunt group supervisor to monitor a call of anyone in the associated hunt group.
Station Speed Dial Station Speed Dial Programming	382 383	Dials/programs one of the 10 Station Speed Dial numbers when followed by a location code (0–9). 86xx phones use the Station Speed Dial button plus a location code for programming and dialing, or they can program Speed Dial buttons for one-button dialing.
Steal	387	Allows the supervisor to take away a call from the hunt group member/agent.
Switch Keymap	399	*(Not used on single line phones)* Allows a phone user to switch between standard and alternate keymaps.

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Table 131:   Phone Feature Codes (continued)

FEATURE NAME	CODE	DEFINITION
System Forward Enable System Forward Disable System Forward On/Off	352 353 354	Enables or disables the database-programmed System Forwarding feature for this phone. The on/off feature code can be used to toggle the feature on or off.
System Speed Dial	381	Dials one of the 1000 System Speed Dial phone numbers when followed by a location code (000–999).
Transfer to Hold	346	Transfers a call to another phone and places it on individual hold so that it does not ring or send call waiting signals until it recalls.
Transfer to Ring	345	Transfers a call to another phone or to an outside phone number.

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SIP AND ITP DEFAULT FEATURE CODES

The following tables show default feature codes for SIP and ITP (Inter-Tel Protocol) Mode phones. For more information about these features, refer to the applicable phone user guide.

SHOW IP FEATURE CODE

The Show IP feature code displays different information in SIP and ITP modes.

Table 132:   SIP and ITP Mode Functions for Show IP Feature

FEATURE	DEFAULT CODE	SIP MODE	ITP MODE
Show IP (or Display Time/Date)	300	Displays the IP address of the phone.	Displays the system date and time, extension number, and status for IP and digital telephones. The IP address is *not* displayed in IP mode.

SIP MODE PHONE FEATURE CODES

Table 133 shows default feature codes when operating in SIP mode.

Table 133:   SIP Default Feature Codes

FEATURE	CODE	FEATURE	CODE
Answer (Ringing Call)	351	Hold – Individual	336
Call Forward All Calls	355	LCD Contrast	303
Conference	5	Message	365
Do-Not-Disturb	370	Microphone Mute On/Off	314
Do-Not-Disturb Cancel	371	Redial	380

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Table 133: SIP Default Feature Codes (continued)

FEATURE	CODE	FEATURE	CODE
Do-Not-Disturb On/Off	372	Redirect Call	331
Group Listen	312	Reverse Transfer (Call Pick-Up)	4
Headset On	315	Ring Tone Selection	398
Headset Off	316	Transfer To Ring	345
Headset On/Off	317		

ITP MODE FEATURE CODES

Table 134 on **page 776** shows default feature codes for 86xx ITP mode phones.

Table 134: ITP IP Default Feature Codes

FEATURE	CODE
Account Code – Following Calls	391
Account Code – Optional	390
ACD Agent Log In	326
ACD Agent Log Out	327
ACD Agent Log In/Out	328
ACD Agent Wrap-Up Terminate	329
Activate Door Relay	332
Agent Help Request	375
Agent Help Reject	376
Answer (Ringing Call)	351
Automatic Intercom Access On/Off	361
Automatic Line Access On/Off	360
Automatic Line Answer	350
Background Music On/Off	313
Call Forward All Calls	355
Call Forward If Busy	357
Call Forward If No Answer	356
Call Forward No Answer/Busy	358

Table 134:   ITP IP Default Feature Codes

FEATURE	CODE
Conference	5
Default Station	394
Directory	307
Display Time And Date	300
Do-Not-Disturb	370
Do-Not-Disturb Cancel	371
Do-Not-Disturb On/Off	372
Do-Not-Disturb Override	373
Dynamic Extension Express On	363
Dynamic Extension Express Off	362
Dynamic Extension Express On/Off	364
Dynamic Extension Express Handoff	388
Feature Button Default	395
Group Listen	312
Handsfree On/Off	319
Headset On	315
Headset Off	316
Headset On/Off	317
Hold – Individual	336
Hot Desk On/Off	348
Hold – System	335
Hunt Group Remove	322
Hunt Group Replace	323
Hunt Group Remove/Replace	324
LCD Contrast Control	303
Message	365
Message – Cancel Message Left	366
Message – Cancel Message on Phone	368

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Table 134: ITP IP Default Feature Codes

FEATURE	CODE
Message – Silent Message	367
Microphone Mute On/Off	314
Page	7
Page Receive On/Off	325
Program Buttons	397
Program Passcode	392
Queue (Callback) Request	6
Record-A-Call	385
Redial	380
Redirect Call	331
Reminder Message	305
Reminder Message Cancel	306
Remote Programming	359
Reverse Transfer (Call Pick-Up)	4
Review Buttons	396
Ring Intercom Always On/Off	377
Ring Tone Selection	398
Routing Off	304
Station Speed Dial	382
Station Speed Dial Programming	383
System Forward Enable	352
System Forward Disable	353
System Forward On/Off	354
System Speed Dial	381
Switch Keymap	399
Transfer To Hold	346

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ADMINISTRATOR FEATURE CODES

Table 135 summarizes the default feature codes for system administrator phones. For feature descriptions, refer to the following guides:

- MiVoice Office 250 Phone Administrator Guide, part number 580.8001

- MiVoice Office 250 Unified Voice Messaging Administrator Guide, part number 580.8009.

Table 135:   System Administrator Default Feature Codes

FEATURE	CODE [EUROPE]
Automatic Diagnostics Delivery On/Off	9823
Clear Network Alarm	9851
Clear System Alarm	9850
Compression On/Off	9982 [9182]
Compression Statistics	9981 [9181]
Diagnostics On/Off	9900 [9100]
Enable Network Day	9862
Enable Network Night	9861
Modem Disable	9867
Modem Enable	9866
Modem Reset	9869
Night Ring On/Off	9860
Periodic Diagnostics On/Off	9825
Program Database	9932 [9132]
Program System Speed Dial	9801
Set Network Date/Time	9810
Set Time/Date	9800
Synchronize Network Time	9811

DIAGNOSTICS MODE FEATURE CODES

The Diagnostics Mode feature code 9900 [9100 in Europe] must be entered before the applicable feature codes can be used.

Table 136:   Diagnostics Mode Default Feature Codes

DIAGNOSTIC FEATURE	CODE U.S.	CODE EUROPE/ AUSTRALIA
ASAI Snoop Off	9926	9126
ASAI Snoop On	9927	9127
Heap Dump	9943	9143
Dump Extension	9933	9133
Dump Node Information	9936	9136
Heap Statistics	9947	9147
ISDN View	9948	9148
Major Reset	9962	9162
Mark As Leaks	9945	9145
Mark As Quiescent	9946	9146
Minor Reset	9964	9164
Network Freeze Zone, System Histories	9939	9139
Network Unfreeze Zone, System Histories	9989	9189
Network Groups	9963	9163
Print Auxdata	9972	9172
Print Message Log	9975	9175
Print Network Log	9976	9176
Query Node Traffic	9978	9178
Show Version	9928	9128
SIP View	9987	9187
Spare 1–3	9910–9912	9110–9112
System History	9974	9174
Diagnostic – View Displays	9983	9183
Seize Device	9973	9173
System History – Freeze	9993	9193
System History – Unfreeze	9998	9198

TROUBLESHOOTING SYSTEM-LEVEL ISSUES

Table 137 summarizes the troubleshooting strategies recommended for resolving discrepancies that may occur at the system level.

Table 137:   System-Level Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Repeated occurrence of all calls in progress dropping **Note:** The central office (CO) must provide a minimum of 18 mA loop current.	AC line is not isolated and dedicated	Have isolated, dedicated line installed. For more information, refer to the Installation chapter in the *MiVoice Office 250 Installation Manual* .
	Defective power supply	Replace the chassis if the power supply is faulty.
	Equipment chassis located near a strong magnetic field or other potential source of interference (copy machines, power transformer, and so forth)	Relocate the equipment chassis a minimum of 20 ft. (6 m) from any equipment that is a potential source of interference.
	IC-CO/CO-CO Disconnect timer(s) need(s) adjustment	See the *dialed digits* field in SMDR. Set timer(s) to a higher value. See "Timers and Limits" on **page 756**.
	Defective Processor Module	Replace the module if faulty.
	Open or loose connection in the cable between the power supply and the chassis backplane, or a defective cable	Turn off the AC power. Check to see that the backplane-to-power supply interface cable is properly connected. Repair or replace the cable and/or the backplane if the connection is faulty. Repair or replace the chassis if the power supply is faulty.
All phones in the system are inoperative; no LED indication when a trunk or call button is pressed	Defective power supply or connector	Replace the chassis.
	Defective module	Replace the faulty module.
	Defective chassis backplane	Check the system voltages on the chassis backplane. Replace the chassis assembly or the chassis backplane if necessary.
Database restore aborts before finishing	Computer's power saver feature shuts down the computer and severs the communications link	Disable the computer power saver feature.

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Table 137:   System-Level Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
DISA does not work	User error	For procedure instructions, see "Using DISA" on **page 181**.
	Phone not compatible	User must dial in from a DTMF telephone.
	Programming error	Ensure that the trunk group is identified correctly as a day or night DISA trunk group. See "Trunk Programming Options" on **page 236**.
	Defective trunk module	Replace the associated module.
	Defective processor module	Replace the module if faulty.

Note: Due to the natural characteristics of the CO trunk, the volume level of DTMF tones transmitted over the trunk may be substantially reduced before reaching the system. This natural degradation in tone volume may adversely affect the reliability of the DISA feature. Other factors which can affect DISA performance are CO trunk noise and the quality and strength of the DTMF tones generated by the off-premises phone itself.

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Unable to interface with computer call-up device (banking machine, answering machine, auto dialer, and so forth)	Equipment being called is defective	Ensure that the called equipment is functioning correctly.
	DTMF digit duration/pause specifications of called equipment is incompatible with Mitel system	Check with the equipment manufacturer for DTMF digit duration/pause specifications. Adjust DTMF Digit Duration/Pause timer. The default value is.06 sec. See "Timers and Limits" on **page 756**.
	CO trunk is designated for dial-pulse signaling	CO trunk must be designated as DTMF. See "Programming Trunk Group Options" on **page 506**.
	Defective trunk module	Replace the associated trunk module.
	Defective processor module	Replace the module if faulty.
No Music On Hold/ Background Music (external music source connected)	External music source turned off or inoperative	Check the external music source for proper operation. For more information, see "Music-On-Hold and Background Music" on **page 322**.
	Programming error	Make sure the Music-On-Hold option for the affected device is programmed correctly. See "Music-On-Hold and Background Music" on **page 322**.
	Defective cable between music source and the music on hold port on the back of the chassis	Repair or replace the cable. Check to see that a 1/8-in., 2-conductor (mono for CS5000, stereo or mono for HX Controller), mini-phone plug was used.
	Defective MOH port	Replace the defective chassis.

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Table 137:   System-Level Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
RFI/EMI present over conversations	AC power source or grounding incorrect	Verify that the AC circuit is isolated and dedicated and check for adequate grounding. For more information, refer to the Installation chapter in the *MiVoice Office 250 Installation Manual* .
	Grounding point is source of RFI/EMI	While the system is running on AC power, temporarily remove the grounding wire to see if it is the source of the RFI/EMI. For correct grounding requirements, refer to the Installation chapter in the *MiVoice Office 250 Installation Manual* .
	AC power source is causing RFI/EMI	If an external battery back-up power source is installed, switch system operation to battery back-up power by unplugging the power source's AC power cord (with grounding wire connected to chassis). If RFI/EMI stops, the AC power source is the cause. Install an RFI/EMI filter or equivalent on the AC outlet.
	Trunk from central office is picking up interference	At the CO block, remove the bridging clips for the trunk. On the telco side of the block, use a test set to check for interference. Also, move the CO trunk to a known good CO circuit. If the problem follows the trunk, contact the telco.

Note: For further RFI/EMI troubleshooting assistance *while on site*, certified technicians should contact Technical Support with the following information:
1. Modulation (AM, FM, or other) and frequency of the interfering station (in Hz)
2. Broadcast power and distance between equipment chassis and broadcast antenna
3. Who hears RFI
– Outside call - inside party only?
– Outside call - outside party only?
– Outside call - both parties?
– Intercom call - one or both parties?
4. Type of instrument(s) on which RFI is heard

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Faulty DID numbers displayed at attendant phones	User error (after dialing the correct 3- or 4-digit "base" number, the user entered in correct or incomplete "start" digits)	Only valid DID "start" digits can be processed by the system.
	Programming error	Ensure that all valid DID numbers have been programmed to ring in to the appropriate locations.
	Defective T1M or T1M-2 expansion module	Replace the T1M or T1M-2 expansion module.
	Defective SLA or Digital Endpoint Module	Replace the SLA or Digital Endpoint Module.

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Table 137:   System-Level Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
DID numbers routed before all digits are dialed	Non-programmable system timer expiring before CO completes digit transmission	The system allows seven seconds for DID input. After seven seconds, the system routes the call based on information received to that point. The Central Office must complete DID digit transmission within seven seconds.
The traceroute command always displays "no reply."	The routers or destination host on the network are programmed to ignore ICMP Echo requests or to not send ICMP Time Exceeded messages.	Contact the network administrator to enable these ICMP messages.
With the diagnostics mode enabled, the phone administrator cannot access the dump from his or her phone. The phone administrator sees "Invalid Extension Number."	The phone administrator has not enabled the diagnostics mode.	Make sure the phone administrator has enabled the diagnostics mode (9900 in the U.S. and 9100 in Europe).
	The user has not entered the correct digits.	Verify the correct diagnostics code is used (9933 for U.S., 9133 for Europe). Also, verify that the correct digits are used for the dump.
The administrator phone is not displaying Alarm 128.	The Audio Diagnostics Suppression System flag is enabled.	Disable the Audio Diagnostics Suppression flag (under System\Flags).
Alarm 128 displays on an administrator phone.	A user completed the Audio Diagnostics feature.	Collect the Freeze which contains Message Print entry and other diagnostics data to submit to Technical Support for analysis.

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Chapter 13

USERS

INTRODUCTION

This chapter provides programming instructions for creating Users in the Users folder. You can create a User to group phones, external numbers, and other contact information, and use the User to associate with an end user instead of a phone.

Users also provide the functionality to associate a group of internal and external destinations with a single user for the purpose of routing calls to the user.

NOTE	Throughout this section, when the capitalized term "User" is used, it refers to the Users element in MiVoice Office 250 DB Programming.

Users is a folder, at the System and Voice Processor level, in DB Programming.
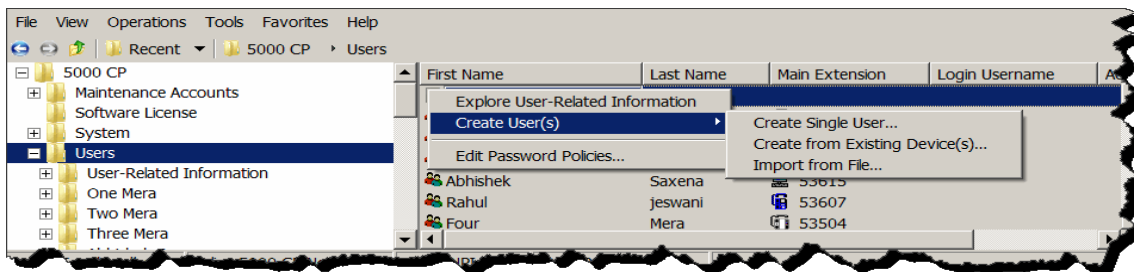
Users provide the following support and functionality:

- **Support for Mitel Application Suite (MAS) Integration**: The Users element tightly couples the Mitel Applications Suite (MAS) with the MiVoice Office 250. MAS currently supports the User concept, and with the addition of the User element in the MiVoice Office 250, MAS users can be imported into the MiVoice Office 250 database and MiVoice Office 250 users can be exported to MAS. See **page 787** for programming information.

- **Support for the Dynamic Extension Express feature**: The Dynamic Extension Express feature includes several fields and options in the Users folder and subfolders. See **page 807** for more information.

- **Support for the Unified Communicator® (UC) Advanced 3.0 or later product**: The UC Advanced product uses the values programmed in the Users Associated Extensions folder and allows the UC Advanced user to activate and de-activate them.

- The ability to create **Administrator** and **Customer Confgurator** user types. See **page 803** for more information.

Configuring the Users programming fields in DB Programming is only required if the MiVoice Office 250 site is using the Dynamic Extension Express feature or a MAS server (for example, MiCollab Unified Messaging). The function of this programming area will be enhanced in future releases.

The Users concept has been incorporated into System Open Architecture Interface (OAI) with the addition of one new event, four new commands and several event and command enhancements.

USER PROGRAMMING

The Users folder groups phones, external numbers, and other contact information, so that it can be associated with an end user instead of a phone. Users also provide the functionality to associate a group of internal and external destinations with a single user for the purpose of routing calls to the user.

To program Dynamic Extension Express (see **page 807**), Users must be created under the Users folder in DB Programming. A User provides the system a way to associate phones, external numbers, and other information to an end-user instead of a phone. Users have a Main extension/Desk destination, other associated destinations (Voice Mail, Mobile Number, etc.), routing steps, and attributes such as:

- **First Name**: Indicates the User's first name.

- **Last Name**: Indicates the User's last name.

- **Mail Extension**: Indicates the extension shown in the directory for the User. This is the device the User is known by.

- **Login Username**: Indicates the login user name to access MAS, the User Web Portal (UWP), and the Administrator Web Portal (AWP). The username can contain 2–21 characters of all non-control characters except the @ or space. The default is blank.

- **Access**: Indicates the type of system access privilege the user has. Users can have one of the following types of access privileges:

  - *User*: Users have a main extension on the system and can access the User Web Portal for account and phone settings.

  - *Customer Configurator*: The Customer Configurator user type may have an extension on the system, will have selected access to Database Programming, and can have access to the User Web Portal for account, phone, and system settings.

  - *Administrator*: The Administrator user type will have a main extension on the system, will have full access to Database Programming, can use System Administration & Diagnostics, and can access the User Web Portal for account, phone, and advanced system settings such as upgrades.
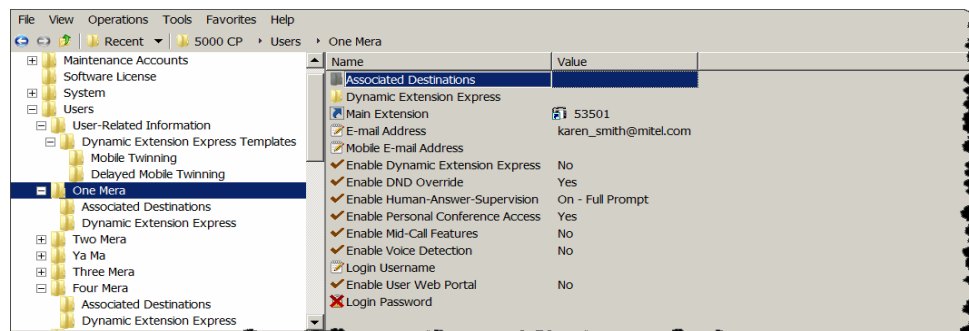
USER PROGRAMMING FIELDS

Table 138 shows fields and folders that are provided for each User.

Table 138: User Fields

FIELD/FOLDER NAME	DESCRIPTION	RANGE/DEFAULT
Associated Destinations	See **page 820**.	
Dynamic Extension Express	See **page 807**.	
Administrator-Related Information	See **page 833**	
Main Extension	Indicates the extension shown in the directory for the user. This is the device the User is known by. After the Main Extension is programmed, the Desk and Voice Mail destinations in the Associated Destinations field (see **page 820**) are auto-programmed for the User corresponding to the Main Extension.	**Range**: Any existing device extension of permitted device type **Default**: None
E-mail Address	Indicates the e-mail address of the User. This address is used by the Configuration Assistant application to send a list of existing Meet-Me Conferencing access codes to the User upon request. If the Mitel Applications Suite (MAS) server is installed, it should be synchronized with the MAS server manually or using the export/import feature.	**Range**: Up to 127 characters **Default**: Blank
Mobile E-mail Address	Indicates the mobile e-mail address of the User. This address is used by the Dynamic Extension Express application to send a text message to the User's mobile phone whenever the User's main extension receives a new station message or voice mail message. See **page 817** for details.	**Range**: Up to 127 characters **Default**: Blank
Enable Dynamic Extension Express	Indicates whether the User has the Dynamic Extension Express feature enabled or disabled. End users can control this flag using a feature code (see "Dynamic Extension Express On" on **page 771**).	**Range**: Yes/No **Default**: No
Enable DND Override	Indicates whether a call to the User's main extension, which is in DND, routes through Dynamic Extension Express or returns DND.	**Range**: Yes/No **Default**: Yes
Enable Human-Answer-Supervision	Indicates whether the User wants to use the Human-Answer-Supervision feature (see **page 813**) when answering a Dynamic Extension Express call through their mobile phone.	**Range**: On - Full Prompt, On - Tone, Off **Default**: On - Full Prompt

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Table 138: User Fields (continued)

FIELD/FOLDER NAME	DESCRIPTION	RANGE/DEFAULT
Enable Personal Conference Access	Indicates if the user has a meet-me-conference with an access code of their main extension.	**Range**: Yes/No **Default**: Yes
Enable Mid-Call Features	Indicates if the user can use mid-call features[1] when using Dynamic Extension Express.	**Range**: Yes/No **Default**: No
Enable User Web Portal	Indicates if the User has access the User Web Portal.	**Range**: Yes/No **Default**: No
Enable Voice Detection	Determines whether or not the Human-Answer-Supervision prompt starts playing when the system detects voice (Yes) or when the system detects that the call has been answered (No). This flag can be set to "Yes" on systems that use trunks that do not have answer-supervision to force the system to detect voice before playing the Human-Answer-Supervision prompt so that the end user always hears the beginning of the prompt. This flag can be set to "No" on systems that are having problems detecting voice during Human-Answer-Supervision. Note that DTMF detection is enabled regardless of whether the system is playing the Human-Answer-Supervision prompt.	**Range**: Yes/No **Default**: No

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1. Mid-Call Featuresare only supported with other networked MiVoice Office 250s running Release v6.0 SP1 and higher software.

This section contains the following information:

- **Creating Users** below

- Deleting Users on **page 791**

- Editing Password Policies on **page 791**

- Description Parsing Rules on **page 791**

- Auto-Creating Users when Phones/Phantoms are Created Manually on **page 792**

- Auto-Creating Users when Phones are Imported from a CSV File on **page 793**

- Creating and Importing Users from CSV Files on **page 794**

- Creating a CSV File on **page 795**

- Importing User Entries from CSV Files on **page 796**

- Batch Creating Users from Existing Phones/Phantoms on **page 801**

- Batch Creating Users from Existing Phones on **page 801**

- Batch Creating Users from Existing Phantoms on **page 802**

- Creating Users Manually in the Users Folder on **page 803**

- Exporting Users on **page 803**

- Deleting Users on **page 804**

- Diagnostics on **page 805**

CREATING USERS

Users are created from the Create User Wizard, which appears for each extenstion created using the Configuration Wizard. It allows you to program the user infromation for each extension individually.

The Create User Wizard is also accessible from Database Programming by right-clicking in the Users folder and selecting **Create Single User**. The Create User Wizard contains the following tabs in its interface:

- User Type

- General

- Destinations

- Dynamic Extension

- Mailbox

See the MiVoice Office 250 Database Programming Online Help for details about using the Create User Wizard.

There are other ways to create Users. The following two methods save the most time and prevent errors in entering duplicate information:

- Auto-create Users when:

  - phones/phantoms are created manually on **page 792**

  - phones/phantoms are imported from a CSV file on **page 793**

- Create and import Users from a CSV file on **page 794**

You can also use the following methods for creating Users:

- Batch create Users from:

  - existing phones/phantoms on **page 801**

  - existing phones on **page 801**

  - existing phantoms on **page 802**

- Create Users manually in the Users folder on **page 803**

- Create and export Users to a CSV file on **page 803**

When a User's folder is presented, DB Programming checks to see if the E-mail System (under System\E-mail Gateway) is set to SMTP with the E-mail SMTP Server programmed. If not, the E-mail Address and Mobile E-mail Address fields appear with a red "X."

In addition, if you programmed an E-mail Address or Mobile E-mail Address for a User, DB Programming checks to see if the E-mail System is set to SMTP (under System\E-mail

Gateway). If it is not, or the E-mail SMTP Server is not programmed, the following warning message appears: "The E-mail system is not currently set up. The [Mobile] E-mail Address will not be utilized until the E-mail system is programmed." Click **OK**, and then you may program the E-mail System.

Additionally, if the Mobile E-mail Address is changed to a non-empty string, DB Programming looks up the mobile destination for the User, and then checks to see if MWI is enabled. If not, the following warning appears: "MWI is not enabled for the mobile destination for this User. The Mobile E-mail Address will not be utilized until MWI is enabled." Click **OK**, and then you may enable MWI for the Mobile destination in the Users\\<*User*>\\Associated Destinations folder.

DELETING USERS

There are several ways to delete Users:

- Delete Users in the Users folder on **page 804**

- Delete Users when Main Extensions are deleted on **page 804**

EDITING PASSWORD POLICIES

See "Maintenance Accounts" on **page 48** for details.

DESCRIPTION PARSING RULES

DB Programming uses phone/phantom Descriptions to auto-program a User's First Name and Last Name fields (up to 20 alphabetical characters) when a User is created from a phone/phantom. The Description field is parsed (resolved) into the Last Name and First Name fields under the top-level Users folder according to the following rules (these rules are the same as the phone import/export logic for description parsing):

- If a comma exists in the Description field, the characters preceding the comma are parsed for the Last Name and those following for the First Name.

- If no comma exists, but a space exists, the characters preceding the space become the First Name and those following the Last Name.

- Otherwise, the entire description is considered the First Name.

To create a User and apply the Description Parsing Rules:

1. Select System – Devices and Feature Codes – **Phones**.

2. Create phones, and type their descriptions.

3. Select phones, and then right-click **Create Users for Selected Phones**. The descriptions are also-programmed in the Users folder.
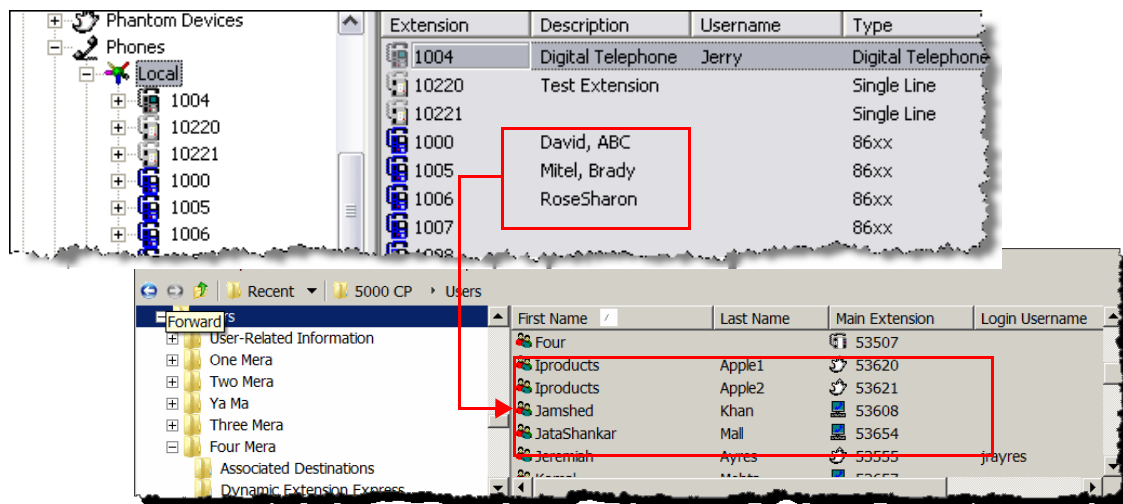
AUTO-CREATING USERS WHEN PHONES/PHANTOMS ARE CREATED MANUALLY

When phones/phantoms are created, a "Create User" flag appears on the dialog in a similar fashion to phones. After the User is created, the phone/phantom becomes the Main Extension and Desk destination for the User and the Main Extension's Voice Mail becomes the User's Voice Mail destination.

To auto-create Users:

1. Do one of the following:

   Create Users for IP Phones:

   a. Do one of the following:

      - Select **System – Devices and Feature Codes – Phones**. Right-click anywhere in the right screen.

      - Select **Tools – Configuration Wizard**. From the IP Device Setup page, right-click and select **Create IP Device**.

   b. Select the phone type that you want to create.

   Create Users for Single Lines:

   a. Do one of the following:

      - Select **System – Controller –** *<Bay#: Single Line Module 2/4>*.

      - Select **System – Controller – Digital Expansion Interfaces – *<Digital Expansion Interface#> – <Bay#: Single Line Module - 8>***.

      - Select **Tools – Configuration Wizard**. From the **Single Line Module Circuit - 4/8:** Configuration, select the port, and then click **Change**.

   b. Select **Single Line**.

Create Users for Digital Telephones:

a. Do one of the following:

- Select **System – Controller – Digital Expansion Interfaces – <Digital Expansion Interface#> – <Bay#: Digital Telephone Module - 16>**.

- Select **Tools – Configuration Wizard**. From the **Digital Endpoint Module - 16: Configuration**, select the port, and then click **Change**.

b. Select **Digital Telephone**, **Digital Telephone/MDPM**, **Dual Single Lines**, or **Single Line/DID**.

Create Users for Phantoms:

a. Select **System – Devices and Feature Codes – Phantoms & Hot Desk Profiles**.

b. Select **Create Phantoms**.

The Create Extension or Batch Create dialog box, similar to the one below, appears.
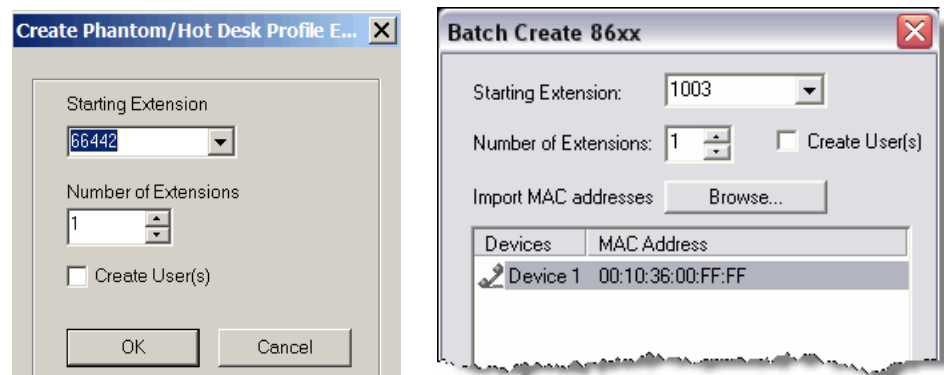
2. Select **Create User(s)**, and then click **OK**. A User is created in the Users folder.

AUTO-CREATING USERS WHEN PHONES ARE IMPORTED FROM A CSV FILE

The Import Phones dialog box has been enhanced so that User information is imported along with the phone information from a file if the Create User field is set to **Yes**. After the User is created, the phone becomes the Main Extension and Desk destination for the User.

The following fields from the import file are programmed automatically for the User:

- Last Name

- First Name

- Email Address

- Login Username

To auto-create Users when phones are imported:

1. Do one of the following:

Method A:

    a. Select System – Device and Feature Codes – Phones – **Local**.

    b. Right-click the right page and select **Create IP Phones from File**.

    Method B:

    a. Select Tools – **Configuration Wizard**.

    b. From the IP Device Setup page, right-click and select **Create IP Phones from File**.

    The Create Phones From File dialog box appears.
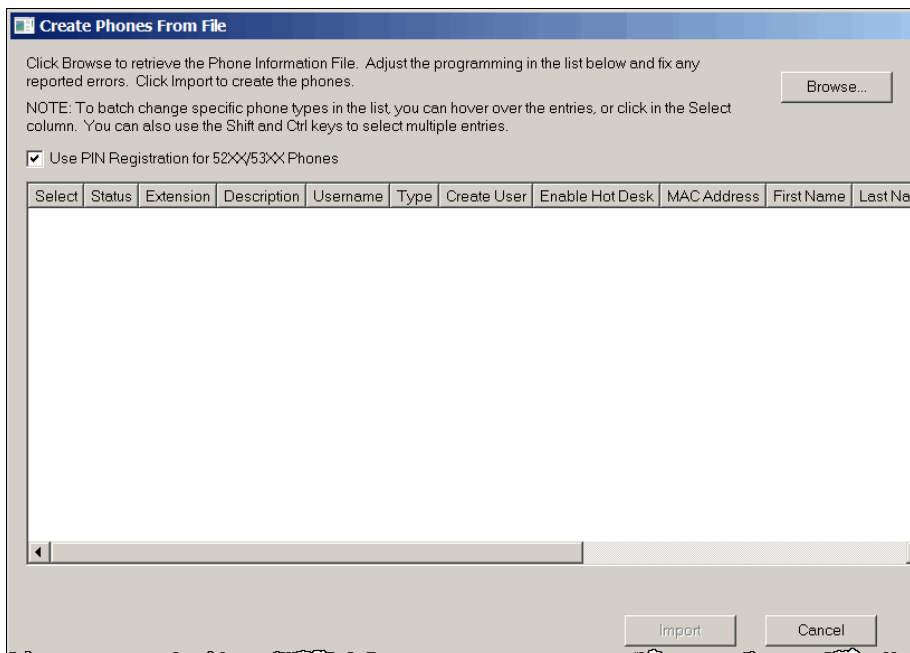
2. Select the phone, and then select **Yes** in the Create User field.

3. Click **Import**.

CREATING AND IMPORTING USERS FROM CSV FILES

You can create User entries in a CSV file and import/export the entries from/to DB Programming.

Considerations when Creating a CSV File

A CSV file is an industry-standard format for text files containing data fields delimited by commas. DB Programming expects a file to use either the TXT or CSV file extension and adhere to the other properties of a CSV file. Remember the following when creating a CSV file:

- Each line in the file represents information for a single item. Information cannot be continued from one line to another.

- Data fields are delimited by commas.

- Only printable characters are considered part of a data field. Control characters within a data field are ignored.

- Each line contains the same number of data fields.

- If a comma is to be considered part of the data, that entire data field must be escaped by double-quotes at the beginning and end of the data field (for example, the data: Jones, Jim would be represented by the data field: "Jones, Jim").

- If a double-quote character is to be considered part of the data in a field, the entire field must be escaped by double-quotes at the beginning and end of the field and the double-quote character itself must be escaped by a preceding double quote (for example, the data: Sara "S" would be represented by the data field: "Sara ""S""").

- Regarding headers:

  - To use user-defined headers in a CSV file, the headers must be listed on the very first line. DB Programming reads the first line of the file and if no digits are identified, the first line is assumed to contain headers.

  - If the first line is identified as a header by DB Programming, DB Programming reads each field and tries to match the field with a predefined attribute.

    Headers are used when parsing the remaining fields in the CSV file. The order of the headers corresponds to the order of the fields in the remaining entries of the file. As DB Programming parses each field, it uses the header order to determine which type of attribute it is reading. For example, if "LAST_NAME" was the first data field read, the first data field of each successive line in the file is considered a last name.

- Additional fields and headers not associated with the predefined attributes are ignored.

CREATING A CSV FILE

The following instructions describe how to create a User file.

To create User entries in a file:

1. Open the template file, `Endpoints`, located in C:\Users\Public\Documents\5000CP\Templates or C:\InterTel\CS5000\Templates (if the system was updated from v3.0 or earlier) on the DB Programming computer.

2. Type the information that you want to import. The fields include:

   - **LAST_NAME**: Last name of the User.

   - **FIRST_NAME**: First name of the User.

   - **EXT_PHONE** (Main Extension): Main Extension of the User. This also becomes the Desk destination for the User (see **page 821**).

   - **EMAIL_ADDRESS**: E-mail address of the User to be used by MAS.

   - **LOGIN**: Login of the User to be used by MAS.

   The following is an example of the User information listed in a Microsoft® Excel® spreadsheet.
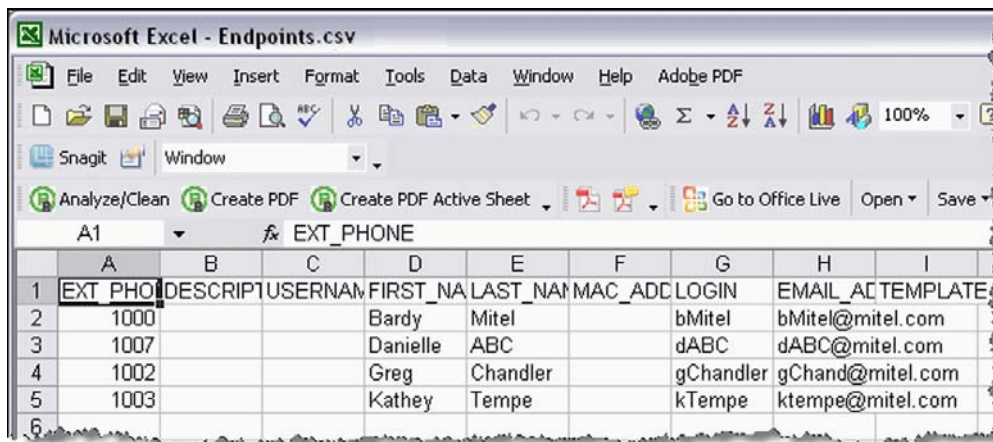
3. Save the data as a CSV file. The following is an example of the file data after it is saved.

IMPORTING USER ENTRIES FROM CSV FILES

You can import User entries from CSV files. If any digits are present or any headers are missing in the first line of the file, DB Programming assumes there is no header in the file. The fields are defaulted to blank.

To import Users from a CSV file:

1. Select **Users**.

2. Right-click in the right pane, and then select **Create User(s)→Import from File. . .** . The *Create Users from File* dialog displays.

3. Click **Browse** to navigate to the file being imported.

4. Select the file, and then click **Open**. The file will populate the *Create Users from File* dialog.
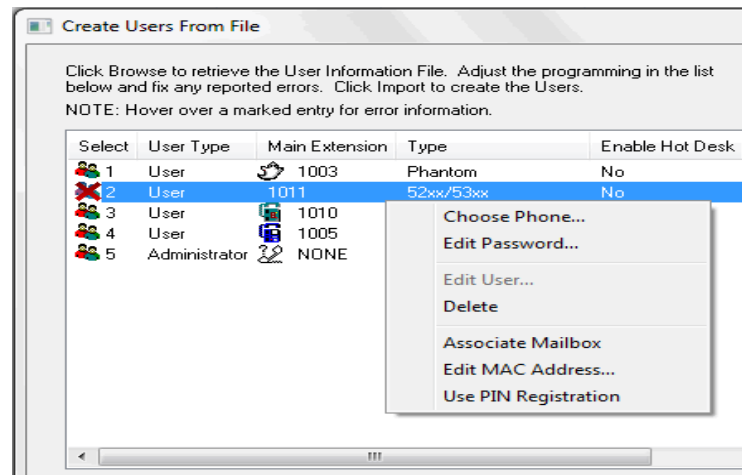
DB Programming will determine which users currently programmed on the system conflict with those being imported. All users on the system must have unique main extensions as well as usernames. Any users that are to be imported that conflict with current users' main extensions or usernames will be marked with a red X to denote that they are invalid and need to be resolved before the import can proceed.

5. You may add, edit, and remove users from the user list during the import.
   To add a user, right-click (in the blank user area) and select the **Add User** option. The create user dialog will be displayed as if creating a single user, but when done the User will be added to the list and not yet created on the system. After the new user has been added it will be validated against the currently programmed users, phones, and features.

   To edit a user in the list, right click on a user and select the **Edit User** option. The create user dialog will be displayed and the fields will be pre-populated with the data retrieved from the file.

   To remove items from the users list, right click on the item and select **Remove User(s)**.

   The edits are *not* seen or saved in the CSV file; they are saved in DB Programming. If DB Programming identifies errors or warnings, a message appears as shown below, and the Import button appears dimmed.
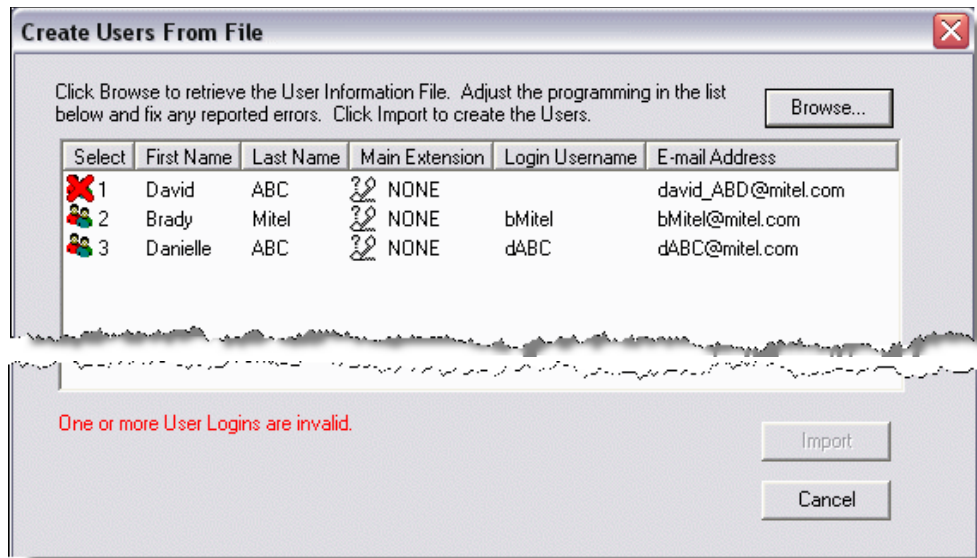
6. After the error(s) are fixed, the Import button becomes visible. Click **Import**. DB Programming allows duplicate entries. It is up to you to review the programming and remove duplicates if desired.

New extensions can be added for the main extension in the template. Since the new extensions can be referenced as a main extension or destination when a user is being edited (with the existing enforcer dialog being used to display them), they must be created on the system right after the data is retrieved from the file.

If any data required to create a new extension is missing, the entry will be marked with a red X and a tool-tip will indicate the error. In addition, the extension will not have an icon next to it because it does not yet exist. Clicking on it will open up a drop-down list with available extensions. There will be a context menu with options to fix specific errors, edit some of the fields, edit or delete the user.

For example, if extension 1011 has a red X on it, and no icon, there are additional options on the last section of the context menu to

• Associate Mailbox

• Edit MAC Address

• Use PIN Registration

This means the extension has not been created yet. The Associate Mailbox option indicates there is a mailbox with the same extension and it can be associated with the new extension when it is created. For this specific phone type, a MAC Address must be entered or the extension can be created by using PIN registration. The extension will be created when all required information is valid.

The following table describes the import fields from the file and the dialog. Some fields have columns in the file but are not displayed on the dialog.

Table 139:   Create User from File Import Fields

IMPORT DIALOG FIELD NAME/ IMPORT FILE COLUMN NAME	VALID VALUES	DESCRIPTION
User Type USER_TYPE	Customer Configurator, Administrator, User	The type of user to create. If the value from the import file is invalid, the user type is defaulted to User.
Main Extension EXT_PHONE	Valid extension of permitted device type	Corresponds to the Main Extension field. An icon next to the extension indicates it exists on the system. If the extension doesn't exist, clicking on it displays a dropdown list with available extensions.
Type PHONE_TYPE	52xx/53xx 86xx Hot Desk Profile IP Single Line Adapter IP Softphone Phantom SIP Phone UC Advanced Softphone	The main extension's phone type. If the extension already exists, this value is ignored and the actual phone type is displayed on the import dialog and cannot be changed. If the value from the import file is invalid, this field will show 'Invalid' and a valid type must be selected. If the extension is NONE, this field will show 'N/A'
Enable Hot Desk ENABLE_HOT_DESK	Yes No	Indicates if Hot Desking for the main extension should be enabled. If the extension already exists, this value is ignored and the actual value is displayed and cannot be changed. N/A will be displayed if this field doesn't apply to the phone type.
Not displayed on the import dialog	MAC_ADDRESS Valid MAC Address	If the value from the import file is invalid and the extension doesn't exist yet and requires a MAC Address, there will be an option on the context menu to edit this value.
First Name FIRST_NAME	20 characters	Corresponds to the 'First Name' field for a user in the normal DPB view.
Last Name LAST_NAME	20 characters	Corresponds to the 'Last Name' field for a user in the normal DPB view.
E-mail Address EMAIL_ADDRESS	Valid E-mail, 127 characters	This field corresponds to the 'E-mail Address' field under the User.
Login Username LOGIN	21 characters	Corresponds to the' Login Username' field for a user in the normal DPB view. Must be unique among all users. This field is required if User Web Portal is enabled.

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Table 139:   Create User from File Import Fields (continued)

IMPORT DIALOG FIELD NAME/ IMPORT FILE COLUMN NAME	VALID VALUES	DESCRIPTION
Login Password WEB_ACCESS_PASSWORD	Valid system passwords	The value must meet the minimum password requirements. This field is required for Administrator and Advanced Administrator user types.
Enable User Web Portal ENABLE_WEB_ACCESS	Yes No	Corresponds to the 'Enable User Web Portal' field under the User. If the value from the import file is invalid, this field is defaulted to true.
Personal Conference Access PERSONAL_CONF_ACCESS	Yes No	Corresponds to the 'Enable Meet-Me Conferencing' field under the User.
Mid-Call Features	Yes No	MID_CALL_FEATURES Corresponds to the 'Enable Mid-Call Features' field under the User.
Not displayed on the import dialog SYSTEM_ADMINISTRATOR	Yes No	Indicates whether a user has system administrator privilege. This field is added to the file when exporting users. If the import file contains this column as well as the USER_TYPE column, this field is ignored and the USER_TYPE column is used, in which case this flag is assigned the default value based on the user type.
Not displayed on the import dialog DATABASE_PROGRAMMING	Yes No	Indicates whether a user has access to Database Programming. This field is added to the file when exporting users. If the import file contains this column as well as the USER_TYPE column, this field is ignored and the USER_TYPE column is used, in which case this flag is assigned the default value based on the user type.
Not displayed on the import dialog WEB_PAGE_DIAGNOSTICS	Yes No	Indicates whether a user has access to the Web Page and System Administration & Diagnostics. This field is added to the file when exporting users. If the import file contains this column as well as the USER_TYPE column, this field is ignored and the USER_TYPE column is used, in which case this flag is assigned the default value based on the user type.

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BATCH CREATING USERS FROM EXISTING PHONES/PHANTOMS

The following instructions describe how to create Users from existing phones/phantoms.

To batch create Users from existing phones/phantoms:

1. Select **Users**.

2. Right-click anywhere in the right pane, and then select **Batch Create Users from Existing Phones/Phantoms**.

3. When the Create User dialog box appears, select types to include, and the click **Next**.

4. Select one or more devices, and then click **Add Items**.

5. Click **Finish**. DB Programming auto-creates Users for the selected devices that are not already used as the Main Extension for any existing Users. The new Users have the associated phone programmed as the Main Extension and Desk destination. The phone/phantom's Voice Mail extension is programmed as the User's Voice Mail destination. The Last Name and First Name fields are programmed based on the device's Description field according to the "Description Parsing Rules" on **page 791**.

NOTE	After the User is created, there is no longer a relationship between the Description field and the Last Name and First Name fields. Any future changes to the Description field will not change the First Name, Last Name fields, and vice versa. This prevents any special description from being overwritten inadvertently.

BATCH CREATING USERS FROM EXISTING PHONES

The following instructions describe how to create Users from existing phones.

1. Select System – Devices and Feature Codes – **Phones**.

2. Select one or more phones.

3. Right-click and select **Create Users for Selected Phones**. DB Programming auto-creates Users for the selected phones that are not already used as the Main Extension for any existing Users. The new Users have the associated phones programmed as the Main Extension, Desk, and Voice Mail destinations. The Last Name and First Name fields are programmed based on the phone's Description field. If any of the selected devices is already programmed as the Main Extension of a User, a warning message appears.

NOTE	After the User is created, there is no longer a relationship between the Description field and the Last Name and First Name fields. This prevents any special description from being overwritten inadvertently.

BATCH CREATING USERS FROM EXISTING PHANTOMS

The following instructions describe how to create Users from existing phantoms.

1. Select System – Devices and Feature Codes – **Phantoms & Hot Desk Profiles**.

2. Select one or more phantoms.

3. Right-click, and then select **Create Users for Selected Hot Desk Profile(s)/Phantom(s)**. DB Programming auto-creates Users for the selected phantoms that are not already used as the Main Extension for any existing Users. The new Users have the associated phantoms programmed as the Main Extension and Desk destination. The phantom's voice mail is programmed as the User's Voice Mail destination. The Last Name and First Name fields are programmed based on the phantom's Description field. If any of the selected phantoms is already programmed as the Main Extension of a User, a warning message appears.

> **NOTE** After the User is created, there is no longer a relationship between the Description field and the Last Name and First Name fields. This prevents any special description from being overwritten inadvertently.
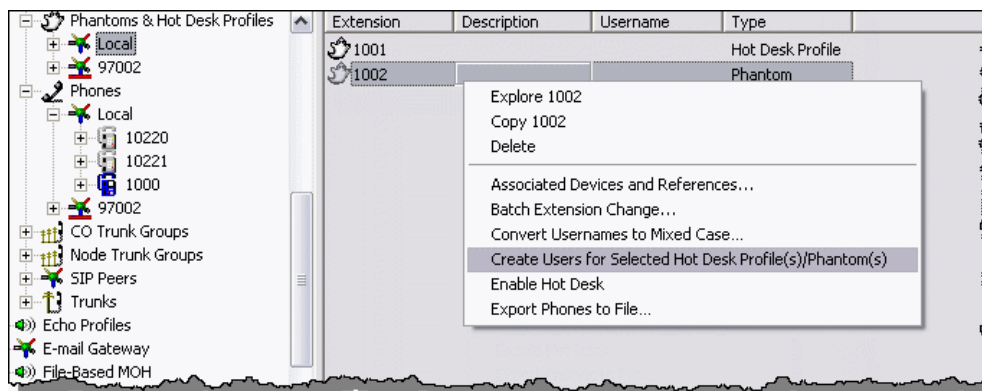
CREATING USERS MANUALLY IN THE USERS FOLDER

The following instructions describe how to create Users manually.

To create Users manually in the Users folder:

1. Select **Users**.

2. Right-click anywhere in the right pane.

3. Select **Create User(s) -> Create Single User**. The Create User wizard appears.

4. Select one of the following type of users:

   • **User**:
   Users have a main extension on the system and can access the User Web Portal for account and phone settings.

   • **Customer Configurator**:
   May have a main extension on the system, will have selected access to Database Programming, and can access the User Web Portal for account, phone, and system settings.
   It will be necessary to click **Details** to set the access rights for this account.

   • **Administrator**:
   Will have a main extension on the system, will have full access to Database Programming, can use System Administration and Diagnostics, and can access the User Web Portal for for account, phone, and advanced system settings such as upgrades

5. Complete the required fields on the **General** tab.

6. Complete the rest of the wizard as necessary.

7. Click **Save**.

EXPORTING USERS

User entries may be exported to a CSV file in a similar fashion to importing the User entries.

To export Users to a CSV file:

1. Select **Users**.

2. Select the items that you want to export.

3. Right-click, and then select **Export User(s) to File**.
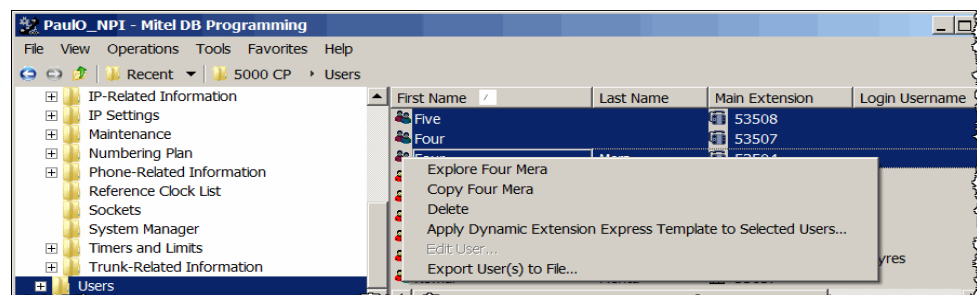
4. When the Export Users to File dialog box appears, rename the file name, and then click **Export**. By default, the file is saved in the Public Documents folder of the user logged in to the DB Programming computer.
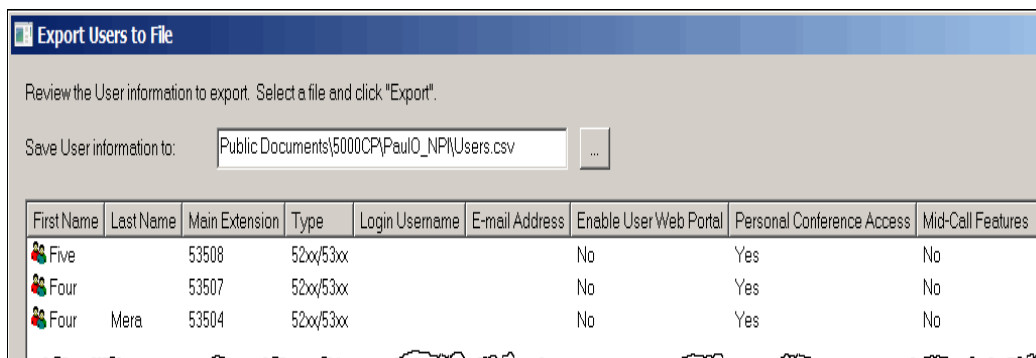
5. If the specified file already exists, DB Programming asks you whether you want to overwrite it. Click **Yes** to overwrite it or **No** to cancel the export.

DELETING USERS

There are two ways to delete Users.

To delete Users in the Users folder:

1. Select **Users**.

2. Select the User(s) that you want to delete.

3. Right-click, and then select **Delete**.

To delete Users when Main Extensions are deleted:

1. Select System – Devices and Feature Codes – **Phones** or **Phantom Devices**.

2. Select the phone(s) or phantom(s) that you want to delete.

3. Right-click, and then select **Delete**. If the phone or phantom is happens to be the Main Extension for a User, the following message appears: "One or more of the devices you are deleting is programmed as the Main Extension for a User. Would you like to delete the User(s) as well?"

4. Click **Yes** to delete Users or **No** to set the Main Extension and Desk destination for the User to "None." The Voice Mail destination remains the same.

DIAGNOSTICS

This section details the diagnostic tools available for Users.

SYSTEM ADMINISTRATION AND DIAGNOSTICS

You can use the System Administration and Diagnostics utility (see page **page 19** for additional information) to access the following dumps:

- **Dump User**: You can dump the User information (System Monitor\Dump User) for all Users or a specific User based on the User's main extension.

- **Dump Extension**: User information has been added to this existing dump. You can dump a station that is associated with a User to see the User information. You can also dump the Personal Router Device (PP065). This provides information about the router device and the calls that are currently being routed.

INFORMATION DUMPS

You can use an administrator phone and the diagnostics feature code 9900 to dump the following information:

- **Dump User (USERS)**: You can see User information for all Users or a specific User based on the User's main extension.

- **Dump Extension (9933)**: User information has been added to this existing dump. You can dump a station that is associated with a User to see the User information.

USER-RELATED INFORMATION

The User-Related Information folder contains the following information:

- Mid-Call Activation/Cancel Code. When the 5000 places a call via Dynamic Extension Express (DEE), this field indicates the feature code for activating or cancelling a Mid-Call Feature (MCF) such as:

  - Hold

  - Transfer

  - Conference Calls

  Two types of conferences are supported:

  - *Basic* – Limited to a maximum of four participants.

  - *Advanced* – Can add up to 20 participants, with limitations the same as those set up through the desktop phone.

  The default feature code is ***#**. See the *MiVoice Office 250 DB Programming Help* for detaiils.

- Mid-Call Confirmation Code. When the 5000 places a call via Dynamic Extension Express (DEE), this field indicates the feature code for confirming or completing a Mid-Call Feature (MCF). The default feature code is ##. See the *MiVoice Office 250 DB Programming Help* for detaiils.

NOTE	Mid-Call Features are only supported with other networked MiVoice Office 250s running Release v6.0 SP1 and higher software.

- **Station Message Callback Number**: Indicates the system-wide number that contains the DID number of the Auto Attendant application. This number is used to support Mobile Message Waiting Indication (MWI). For details about programming Mobile MWI, see **page 817**.

- **Voice Mail Message Callback Number**: Indicates the system-wide number that contains the DID number of the Voice Mail Message Notification and Retrieval application. This number is used to support Mobile MWI. For details about programming Mobile MWI, see **page 817**.

- **Dynamic Extension Express Templates**: See **page 818** for details.

DYNAMIC EXTENSION EXPRESS

This section details the following for the Dynamic Extension Express feature:

• **Feature Description** below

• Functional Behavior on **page 810**

• Programming Dynamic Extension Express on **page 816**

• Troubleshooting and Tuning on **page 823**

• Diagnostics on **page 828**

• Mid-Call Features

> **NOTE** Mid-Call Features are only supported with other networked MiVoice Office 250s running Release v6.0 SP1 and higher software.

For instructions for using the Dynamic Extension Express and Handoff features, refer to the appropriate user guides.

> **NOTE** To use Dynamic Extension Express functionality in a multi-node network, all nodes in the network must be upgraded to v3.2 or later (MiVoice Office 250) to ensure compatibility between nodes.

Software release 5.0 and later includes a number of enhancements to the existing Dynamic Extension Express (DEE) feature, including:

• handoff push/pull from other associated destinations (see **page 808**)

• message waiting indications on all internal associated destinations (see **page 808**)

• additional associated destinations (see **page 816**)

FEATURE DESCRIPTION

The Dynamic Extension Express feature is closely related to the User element in the MiVoice Office 250 database (see **page 787**). Throughout this section, when the capitalized term "User" is used, it refers to the User element in the MiVoice Office 250 database. Dynamic Extension Express is a feature where incoming calls to users are routed to multiple destinations (for example, the user's desk phone, mobile phone, or both at the same time). Ringing multiple destinations at the same time is called twinning.

For software release 5.1 and later, all Dynamic Extension Express features, except for Handoff-Push, are supported on SIP phones.

Handoff Features

Dynamic Extension Express also includes the following Handoff features:

• **Handoff Push**: Users can "push" (move) an active call from any of their internal associated destinations (see **page 808**) to an associated destination without disconnecting audio. The push function is useful if the user is leaving the work area, but wants to remain connected to an active call. The push function rings all associated destinations (except voice mail) and the user selects which device to use to answer the call. When the user answers the

call, he or she is immediately connected to the other parties at the new destination. When the call is switching over to the new destination, there may be a momentary break in audio, which may be noticeable to the other parties on the call.

- **Handoff Pull**: Users can "pull" a call that was routed to their associated destination, such as a mobile phone) back to any of their internal associated destinations (see **page 808**). This feature is useful when users answer a call on their mobile phone and then return to their desk or other destination. After users pull a call back to their new destination, the call is connected immediately and users can then access system features such as conference, hold, and transfer.

NOTE	The Handoff "pull" feature is not available at the desk phone if any of the following are true: • The desk phone is not the main extension of the User. • The trunk used for the outgoing external call is on a MiVoice Office 250 node equipped with software earlier than v3.2.

Additional Handoff Push/Pull Destinations

Prior to v5.0, DEE users could only perform a Handoff "push" or a Handoff "pull" from their DEE main extension (most likely their desk phone). With v5.0 or later, DEE users can perform a Handoff push or a Handoff pull from any of their internal associated destinations as long as:

- they are not a voice mail destination

- they do not reside on a MiVoice Office 250 node older than v5.0

- they are not a 5610 Cordless Handset or UC Express SIP Softphone

If a user's internal associated destination has a button programmed for the Handoff feature, then the button will light if there is a call available to pull.

See **page 817** for programming instructions.

Message Waiting Indication Destinations

With v5.0 or later, all of the internal associated destinations also receive MWIs as long as:

- they are not a voice mail destination

- their MWI flag is set to Yes

So, if the User's main extension receives a voice mail message or a station message, the MWI is propagated to the User's other internal associated destinations. (Note that the Mitel 5610 Cordless Handset and the UC Express SIP Softphone cannot receive propagated station MWIs from the main extension, but they can receive propagated voice mail MWIs.) If any of the internal associated destinations responds to the MWI, then the MWI is automatically cleared on all internal associated destinations. Note that if any destination other than the User's main extension receives an MWI, it will not propagate to the other internal destinations. The MWI has to originate on the User's main extension.

In DB Programming, there is an MWI flag for each of the User's associated destinations that determines whether or not that destination can receive MWIs. Mitel recommends disabling MWI capability for associated destinations that reside on a MiVoice Office 250 node earlier than v5.0, as the feature may not always work properly (e.g., when the MWI is cleared at one destination, it may not always be cleared at the other destinations). Mitel also recommends not using the MWI propagation capability with the legacy Unified Communicator application.

Toggling the MWI flag for the User's main extension does not impact the current MWI state for the various associated destinations; only future MWIs are affected by the MWI flag.

See **page 823** for programming instructions.

Mobile MWI

If a User's associated Mobile destination has the MWI flag set and is configured with a mobile e-mail address, the system sends an e-mail message to the designated address to indicate that the User's main extension has received a new station message or voice mail message. If the User's mobile phone is configured to receive e-mail messages sent to the designated address, the e-mail message should appear as a Short Message Service (SMS) text message on the mobile phone. (This assumes that all E-Mail Gateway settings are properly configured.) Note that the SMS text message is sent only to the Mobile 1 destination and not the Mobile 2 destination, as you cannot designate an e-mail address for the Mobile 2 entry. (Standard text messaging rates may apply.)

See **page 817** for programming instructions.

The message is sent in the supported language configured for the User's main extension. (Japanese is not supported, so English is used in its place.)

Here is an example of an e-mail message generated by a station message:

```
You have a message from <Username> at extension <Extension>.

Callback: <Auto Attendant Number>
```

Here is an example of an e-mail message generated by a voice mail message:

```
You have a new voice mail message for mailbox number <Mailbox Number>.

Callback: <VM Notification and Retrieval Number>
```

The automated attendant and voice mail notification/retrieval numbers displayed in the SMS text messages are DID numbers programmed in the system database. A mobile phone user can use the number to easily call back into the system and speak to the messaging party or retrieve the voice mail message. If the callback numbers are not configured, then the callback line is omitted.

Note that If this mobile MWI feature is used with MiCollab Unified Messaging as the voice mail system, the user only receives an SMS text message for the initial (i.e., first) *new* voice mail message. Therefore, Mitel recommends not using this feature with MiCollab Unified Messaging.

Feature Codes

There are four feature codes for Dynamic Extension Express. See "Dynamic Extension Express On" on **page 771** for feature code definitions.

- Dynamic Extension Express Off (362)
- Dynamic Extension Express On (363)
- Dynamic Extension Express On/Off (364)
- Dynamic Extension Express Handoff (388)

FUNCTIONAL BEHAVIOR

This section provides functional details for Dynamic Extension Express on the MiVoice Office 250.

Licensing

The Dynamic Extension Express feature requires a system license.

Users and Dynamic Extension Express

The top-level Users folder in DB Programming provides a programming interface for the Dynamic Extension Express feature.

The Users folder contains the following fields and options associated with the feature:

- **<user> options**: When you program a User, you can configure the following Dynamic Extension Express options:
  - Enable Dynamic Extension Express
  - Enable DND Overrides Dynamic Extension Express
  - Enable Manual Forwards Override Dynamic Extension Express
  - Enable Human-Answer-Supervision

    See **page 787** for a description of these fields and programming instructions.

- **Associated Destinations**: Dynamic Extension Express relies on the information programmed in the Associated Destinations folder (Users – *<user>* – **Associated Destinations)** to route calls. The Associated Destinations are included in the Routing steps (see **page 820**).

- **Dynamic Extension Express Steps**: When you first create a User, the default Dynamic Extension Express template is applied as the User's routing steps (Users – *<user>* – **Dynamic Extension Express**). A routing step consists of one or more Associated Destinations combined with a timer. You can create several different routing steps. You can add or remove the Desk, Mobile, Home IP, Home, and Softphone destinations, but the Voice Mail destination must be the final destination (see **page 816**).

- **Dynamic Extension Express Templates**: DB Programming provides two templates for Dynamic Extension Express: Mobile Twinning, and Delayed Mobile Twinning (Users – **Dynamic Extension Express Templates**). Mobile Twinning is set as the default template, however, you can change the default template to Delayed Mobile Twinning (see **page 819**).

DYNAMIC EXTENSION EXPRESS EXAMPLES

DB Programming provides two templates to route calls: Mobile Twinning (default), and Delayed Mobile Twinning. A description for how calls are routed when the template is applied appears below.

- **Mobile Twinning (Template)**: This template provides two routing steps:
  - In the first step, calls simultaneously ring the user's desk phone and mobile phone for 24 seconds.

- In the second step, calls go to voice mail.

Table 140: Mobile Twinning (Template)

STEP NUMBER	RING DURATION (SECONDS)	DESTINATION 1	DESTINATION 2
1	24	Desk	Mobile
2	N/A	Voice Mail	

- **Delayed Mobile Twinning (Template)**: This template provides three routing steps:
    - In the first step, calls ring the desk phone 4 seconds.
    - In the second step, calls ring both the desk phone and mobile phone for 24 seconds.
    - In the third step, calls go to voice mail.

Table 141: Delayed Mobile Twinning (Template)

STEP NUMBER	RING DURATION (SECONDS)	DESTINATION 1	DESTINATION 2
1	4	Desk	
2	24	Desk	Mobile
3	N/A	Voice Mail	

The templates can be used, "as is," or they can be modified to suite your needs. This section provides examples of how you can modify the templates to route calls.

- **Short Mobile Twinning**: This example provides three routing steps:
    - In the first step, calls ring both the desk phone and mobile phone for 6 seconds.
    - In the second step, calls ring just the desk phone for 12 seconds.
    - In the third step, calls go to voice mail.

Table 142: Short Mobile Twinning Example

STEP NUMBER	RING DURATION (SECONDS)	DESTINATION 1	DESTINATION 2
1	6	Desk	Mobile
2	12	Desk	
3	N/A	Voice Mail	

- **All-Ring**: This example provides two routing steps:
    - In the first step, calls ring the desk phone, mobile phone, and softphone for 16 seconds.
    - In the second step, calls go to voice mail.

Table 143: All Ring Example

STEP NUMBER	RING DURATION (SECONDS)	DESTINATION 1	DESTINATION 2	DESTINATION 3
1	16	Desk	Mobile	Softphone
2	N/A	Voice Mail		

HOW CALL PROCESSING ROUTES DYNAMIC EXTENSION EXPRESS CALLS

This section details how Call Processing routes a call based on the User's programmed Dynamic Extension Express information.

Dynamic Extension Express functions like a personal Hunt Group for a User, where the members of the hunt group are the User's Associated Destinations. Unlike a traditional hunt group, this special hunt group can serve all Users by changing members on a per-call basis. By combining User destinations and User routing steps, Dynamic Extension Express creates a dynamic hunt group.

When a call rings the main extension for a User, the phone/phantom device with the same extension receives the call. If Dynamic Extension Express is enabled at the phone/phantom, Call Processing uses the information programmed for the User to route the call appropriately. If, for some reason, Call Processing cannot route a call, the call returns to the User's main extension, where it will ring until answered or the caller hangs up.

| NOTE | Phones utilizing Dynamic Extension Express should be configured with at least two call keys. |

Call Processing routes Dynamic Extension Express calls as follows:

- For each routing step, Call Processing recalculates the response to the calling party who receives one of the following responses:
  - *If at least one destination is ringing*, the calling party hears ringback.
  - *If all the destinations are camped on*, then the calling party hears camp on.
  - *If the calling party is camped on, and then one of the destinations starts to ring*, then the calling party hears ringback.
- Call Processing skips invalid routing steps and destinations.
- If every destination in a routing step is in DND, then Call Processing skips the routing step. When the last routing step is reached, Call Processing redirects the call to the main extension.
- If the User's main extension changes or is unequipped while Call Processing is routing a call, then the call is terminated. If this occurs, the User's main extension may return DND to the calling party.
- Call Processing performs call logging after it routes the call as follows:
  - *If the User's main extension is not involved in the routing steps,* then the system does not make an entry in the call logs for that phone.
  - *If a call advances to a new routing step and the new step does not contain one of the devices in the previous step,* then that device will have a missed call entry for that call.
  - *If a call rings multiple destinations in a routing step and one of the destinations answers the call,* then the other devices in the routing step will not have a missed call entry for that call.

RING DURATION

When you make a call to a mobile phone, there is often a delay before the mobile phone rings. The amount of delay varies between mobile service providers, and sometimes varies from call to call. When configuring Dynamic Extension Express Ring Duration remember to allow users an adequate amount of time to accept calls on their mobile phones. The Ring Duration should be "tuned" to match the mobile carrier and the user's preferences. Note that:

- If the timer is too short, the mobile user will not have the opportunity to answer the call before it is routed to voice mail.

- If the timer is too long, callers will hang up instead of leaving a voice mail message.

Ideally, the timer should be set to the minimum amount of time required for the mobile user to press # to accept the call.

HUMAN ANSWER SUPERVISION

When you include a trunk in a routing step, Call Processing cannot determine if the user answered the call or if the call was answered by voice mail. Enabling Human Answer Supervision forces the user to enter a DTMF tone to control the action of the call after it is routed to the public network.

The "Enable Human Answer Supervision" option under Users – *<**user**>* in DB Programming is used to configure Human Answer Supervision for Dynamic Extension Express (see **page 788**). When this option is enabled, the user must press # to accept the call or * to send the call to voice mail. If the user does not enter a DTMF tone, the call is routed back to the user's main extension.

In some situations, it might be appropriate to disable Human Answer Supervision. For example, some phones have security access codes that require several keystrokes before the user can press #. These types of users may be unable to press # to accept the call before Ring Duration sends the call to voice mail.

SMDR

When Human Answer Supervision is enabled (see **page 813**), if the user does not press # to answer a call that was routed to the public network, the SMDR record reflects that fact with a special character (R). When the user presses # to accept the call the system outputs the normal SMDR record.

In response to the Dynamic Extension Express feature, SMDR has a Human-Answer-Supervision Record option that indicates whether or not the system outputs the "R" SMDR record (see **page 820**).

INTEROPERABILITY WITH OTHER FEATURES

The following information explains how Dynamic Extension Express interoperates with other MiVoice Office 250 features:

- **Do-Not-Disturb:** When the user enables DND for his or her main extension, the system follows the "Enable DND Overrides Dynamic Extension Express" option under Users – *<user>* to determine if incoming calls should be routed by Dynamic Extension Express. See **page 788** for more information.

- **Manual Forwarding**: When the user manually forwards his or her main extension, the system follows the "Enable Manual Forwards Overrides Dynamic Extension Express" option under Users – *<user>* in OLM to determine if incoming calls should be routed by Dynamic Extension Express.

- **System Forwarding**: Dynamic Extension Express overrides system forwarding. To use system forwarding, the user must disable Dynamic Extension Express.

- **Camp-ons**: When a user is on a call or when the system is routing a call and a new call rings in, the system also routes the new call. The second call (and all subsequent calls) camp-on to all of the destinations programmed in the user's Dynamic Extension Express steps. This camp-on functionality allows the user to manage all incoming calls (for example, send to destination) instead of camping on to the Dynamic Extension Express device.

- **Queue**: Because of the complexity of multiple statuses and feature functionality, an end-user cannot queue on to a camped-on personal call routing call.

- **Destination extensions with Dynamic Extension Express enabled**: If a call rings into a destination extension that has Dynamic Extension Express enabled, the system does not follow the destination's Dynamic Extension Express.

- **Transfer**: Calls transferred to an extension with Dynamic Extension Express enabled are routed via Dynamic Extension Express. If the transferring party uses transfer-to-hold, then the system converts the transfer to a transfer-to-ring.
  An unanswered transferred call does not ring back at the transferring party's phone. This is because there are no transfer re-call timers registered. It is necessary to configure Voice Mail or attendant's extension as the last routing step for DEE to handle unanswered calls.

- **Conference**: When a conference call rings an extension that has Dynamic Extension Express enabled, the conference call is routed to the appropriate destinations via Dynamic Extension Express.

- **Multiple Ring-In**: When a user's main extension is part of a trunk group multiple ring-in, the CO call to the user does *not* follow Dynamic Extension Express.

- **ISDN PRI Two B-Channel Transfer**: Dynamic Extension Express works with the existing ISDN PRI Two B-Channel Transfer (TBCT) feature on the MiVoice Office 250. If the user with Dynamic Extension Express enabled receives an external call on a PRI trunk, that was answered after being routed to their mobile phone, the TBCT feature executes. The call is transferred off of the MiVoice Office 250, if all TBCT transfer conditions are met.

  Handoff functionality is not available for a call that uses the TBCT feature because the call is no longer managed by the MiVoice Office 250.

- **OAI**: Any third-party Open Architecture Interface (OAI) application can perform an OAI offer command. If call offering is enabled for the OAI, then the system allows the OAI application to route the call first. For example, if a user is using Mitel Unified Communicator (UC) 5000 and Dynamic Extension Express, then UC 5000 has precedence over Dynamic Extension Express. Mitel does not recommend configuring a single user for two types of routing.

FEATURE LIMITATIONS

The following call types do not follow Dynamic Extension Express:

- Hunt group calls

- Hunt group announcement/overflow calls

- Transfer recalls

- Hold recalls

- Conference recalls

- Attendant recalls

INTEROPERABILITY WITH PERIPHERAL PRODUCTS

This section details the interoperability between existing peripheral products and the MiVoice Office 250.

- **Mitel UC 5000**: If Dynamic Extension Express is disabled and UC 5000 is enabled for a user, the current functionality of UC is the same as previous versions. If both Dynamic Extension Express and UC 5000 are enabled, UC 5000 will override Dynamic Extension Express because of the OAI functionality. Mitel does not recommend configuring a single user for two types of routing.

- **Customer Service Manager**: The Dynamic Extension Express feature adds new system OAI events and commands that will be used by a future release of Mitel Customer Service Manager (formerly Contact Center Suite in U.S. and CallView in Europe). If Dynamic Extension Express is disabled, then CSM functions as it did in previous MiVoice Office 250 versions. If CSM v5.0 and earlier is used with Dynamic Extension Express, then some of the CSM reporting, such as trunk usage, may be inaccurate.

- **Voice Processing Systems**: The Dynamic Extension Express feature can be used with the Unified Voice Messaging (UVM) and Mitel MiCollab Unified Messaging voice processing systems. You can configure the user's voice mail destination as a voice mail application from each voice mail system.

- **Mitel Unified Communicator Advanced v3.0**: Unified Communicator (UC) Advanced v3.0 or later requires MiVoice Office 250 v3.2 or later software to support the following:

  - UC Advanced softphone device type

  - System OAI functionality

| **NOTE** | If using a CT Gateway with UC Advanced, you must use CT Gateway version 4.4. |

PROGRAMMING DYNAMIC EXTENSION EXPRESS

You can program Dynamic Extension Express Steps on a per-User basis in DB Programming. Each User has a Dynamic Extension Express subfolder where routing steps may be programmed to specify how calls to the User's Main Extension should be routed. In online mode, if Dynamic Extension Express is not licensed, each step appears with a red "X" over it. Although the template marked as the default template is used to initially program Dynamic Extension Express for every User, you may go into the Dynamic Extension Express folder of any given User and edit the steps.

The Dynamic Extension Express folder consists of the following fields:

- **Step Number**: Step number for the Dynamic Extension Express folder. The range is 1-5; the default value is 1 (or next available consecutive number).

- **Ring Duration**: Indicates the amount of time that each routing step rings before advancing to the next routing step. This timer defaults to the value of the Routing Step Advance Timer in the System/Timers And Limits folder (see **page 820**).

- **Destination**: Each routing step can have up to 10 destinations. Each routing step has three associated destinations by default. A destination is a type of User Associated Destination to ring for this step. The available types are:

> **NOTE** | Voice Mail must be in its own routing step. Also, Unified Communicator Advanced only supports eight destinations.

- Desk*
- Mobile*
- Softphone (such as a UC Advanced Softphone)
- None
- Home IP—For users who work at home with an IP phone that is connected to the phone system (5xxx, 8xxx, IP SLA, softphone, and so forth).
- Home—A home telephone number with area code for users who work at home but do not have a phone that is programmed on the phone system.
- Voice Mail*—A final destination and only available for the last step, and the last step must only have a single destination.
- Desk 2
- Mobile 2
- Softphone 2
- Home IP 2
- Home 2

  * The Desk, Mobile, and Voice Mail types are auto-created for each User and read-only.

To add new steps:

1. Select Users – *<User>* – **Dynamic Extension Express**.

2. Right-click anywhere in the right pane, and then select **Add Routing Step**. A new step is added with defaults for the fields. You can create up to five steps for each Dynamic Extension Express.

3. Right-click a step, and then do any of the following:

   To add another destination to the step:

   > Select **Add Destination**. A new destination is added as Destination X, where X is the total number of destinations for that step and is set to None by default. There can be up to 10 destinations (or eight for Unified Communicator Advanced).

   To edit a destination:

   > Select the field, and then select the destination from the drop-down list. The available types are None, Desk, Mobile, Softphone, Voice Mail, Home IP, Home, Desk 2, Mobile 2, Softphone 2, Home IP 2, and Home 2.

   To delete a destination:

   > Select **Delete Destination**. If the step has only one destination, this option is grayed out because each step has to have at least one destination.

   To delete steps:

   > Select the step(s) that you want to delete, and then select **Delete Selected Routing Step(s)**. The remaining steps will be renumbered accordingly after the step is deleted.

4. *(Optional)* Change the Ring Duration value.

Programming non-main extensions to support the User Handoff Push feature

To allow a non-main extension to use the Handoff Push feature, the phone must be associated with a User. See **page 808** for details.

To associate the phone with a User:

1. Select System – Devices and Feature Codes Phones – Phones – *<phone>* – **Associated Extensions**.

2. In the **Associated User Extension** field, type the User Main Extension that is to be associated with the phone. See **page 394** for details about the Associated User Extension field.

Programming Users for Mobile MWI

For details about Mobile MWI, see **page 809**.

To set the station message callback number for Mobile MWI:

1. Select Users – User-Related Information – **Station Message Callback Number**.

2. Enter a system-wide number that contains the DID number of the Auto Attendant application. The end-users can use this number to call the auto attendant to return the call of a station message. The DID number can be up to 48 characters (0-9, *, #, p, P). Note that this number is used by the mobile phone and not the MiVoice Office 250; therefore, non-digits (e.g., # and P) may not perform the desired outcome.

To set the voice mail message callback number for Mobile MWI:

1. Select Users – User-Related Information – **Voice Mail Message Callback Number**.

2. Enter a system-wide number that contains the DID number of the Voice Mail Message Notification and Retrieval application. The end-users can use this number to call back voice mail to listen to their new message. The DID number can be up to 48 characters (0-9, *, #, p, P).

To set the mobile e-mail address for the User:

1. Select Users – *<User>* – **Mobile E-mail Address**.

2. Type the e-mail address of the User' mobile phone. The address can contain up to 127 characters. (The mobile e-mail address is typically in the format: `number@<carrier address>`. See the specific mobile phone carrier for the proper format.)

Note that the system E-Mail Gateway settings must be properly configured to use this feature. See "E-Mail Gateway Programming Options" on **page 942** for required settings.

EDITING DYNAMIC EXTENSION EXPRESS TEMPLATES

Templates are pre-programmed sets of Dynamic Extension Express Steps. The purpose of templates is to set up generic call routing scenarios that may be applied to multiple users and reduce manual programming. When a Dynamic Extension Express Template is applied to a User, the Dynamic Extension Express Steps of the template are copied and programmed for the User's Dynamic Extension Express.

There are two templates available: Mobile Twinning and Delayed Mobile Twinning. Each template has the same fields as the User Dynamic Extension Express Steps (see "Programming Dynamic Extension Express" on **page 816** for details). You can change the template name, destination type, and Ring Duration values.

Dynamic Extension Express templates include:

- **Mobile Twinning**: This is the default template. Whenever a User is created, the "Mobile Twinning" template is automatically applied to program Dynamic Extension Express steps for that User. DB Programming auto-programs the steps shown in **Table 144** as the "Mobile Twinning" template for all Users.

Table 144: Mobile Twinning Template Steps

STEP	RING DURATION	DESTINATION 1	DESTINATION 2
1	20	Desk	Mobile
2	N/A	Voice Mail	

- **Delayed Mobile Twinning**: The other default template, "Delayed Mobile Twinning," consists of the routing steps shown in **Table 145**.

Table 145:   Delayed Mobile Twinning Template Steps

STEP	RING DURATION	DESTINATION 1	DESTINATION 2
1	4	Desk	
2	20	Desk	Mobile
3	N/A	Voice Mail	

NOTE	Associated destinations must be programmed correctly for Dynamic Extension Express Steps to be executed completely. For example, if a User is set up with the "Mobile Twinning" Dynamic Extension Express Template, but no outside number is programmed for the Mobile destination of the User, the system will ring the Desk destination only and not perform twinning for Step 1. If the associated Voice Mail destination is set to the "None" Device, Step 2 will not be performed. Instead, the system will execute the Step 1 indefinitely.

To change the default template:

1. Select Users – User-Related Information – **Dynamic Extension Express Templates**.
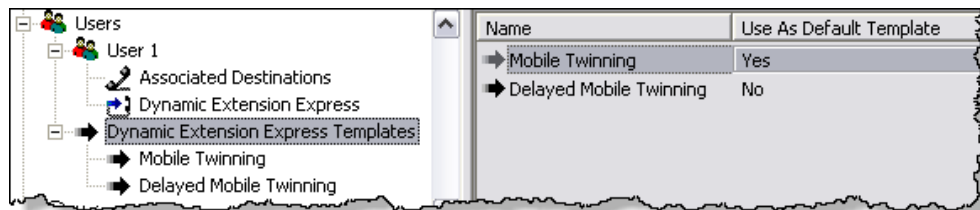
2. Change the Use As Default Template flag of the template that is programmed as the default template from Yes to **No**.

3. Change the Use As Default Template flag of the template you want to use from No to **Yes**. Only one template may have this flag set to **Yes** at a time.

To change the description of the template:

1. Select Users – **Dynamic Extension Express Templates**.

2. Select the **Name** field of the template that you want to edit.

3. Change the description (up to 40 characters).

To use one of the templates on existing Users:

1. Select **Users**.

2. Select one or more Users in the Users folder, and then right-click.

3. Select **Apply Dynamic Extension Express Template**. A selection wizard appears.

4. Select the template to apply to the selected Users. Any Dynamic Extension Express Steps already programmed for the selected Users will be deleted and replaced with the Dynamic Extension Express Steps of the chosen template.
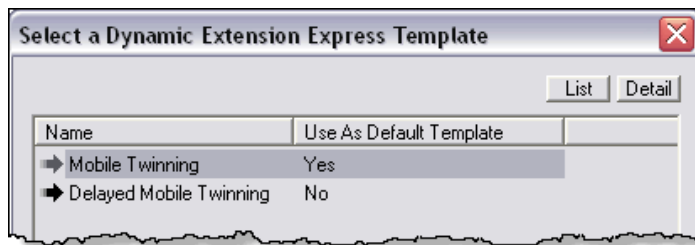
5. Click **OK**.

To change the Ring Duration default value:

1. Select System – **Timers and Limits**.

2. Change the **Routing Step Advance Timer** value. This timer serves as the system-wide default for the Dynamic Extension Express Steps. When a Dynamic Extension Express step is created, this field is used to default the Ring Duration field for the Step. It signifies the amount of time that each routing step rings before advancing to the next routing step. The range is 1-1000; the default is 8 seconds.

RECORDING ALL HUMAN-ANSWER-SUPERVISION CALLS

SMDR has a Human-Answer-Supervision Record option that indicates whether or not the system outputs the "R" SMDR record. See **page 813** for details.

To record all Human-Answer-Supervision calls:

1. Select System – Maintenance – **SMDR**.

2. Select the **Record All Human-Answer-Supervision Calls** flag, and then select the check box.

PROGRAMMING ASSOCIATED DESTINATIONS

Associated destinations are used in conjunction with Dynamic Extension Express Steps for Dynamic Extension Express. For example, when a Dynamic Extension Express Step is programmed to ring the Desk destination of a User, the system routes the call to the User's Desk associated destination. Associated destinations are displayed in the Users\<*User*>\Associated Destinations folder in DB Programming.
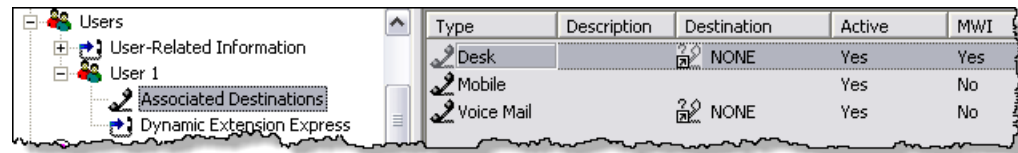
When a User is created, three associated destinations are automatically created for the User: Desk, Mobile, Home IP, Home, Voice Mail, Desk 2, Softphone 2, Home IP 2, Home 2, Mobile 2. If the User was created in a way that the Main Extension was not auto-programmed, these associated destinations have a blank (None) device destination by default.

The total number of Destinations any User may have programmed is 10 (one per Destination Type). UCA only supports up to eight Destinations per User; if you attempt to create the 9th Destination, a warning appears indicating UCA only supports eight Destinations per User.

When the Main Extension for the User is programmed, the Desk destination is programmed with the same device and the Voice Mail destination will be set to the associated Voice Mail extension of the Main Extension.

Because the associated Voice Mail destination of a User must match the associated Voice Mail extension of the Main Extension for the User, if the associated Voice Mail extension of the Main Extension is changed, DB Programming automatically changes the associated Voice Mail destination correspondingly.

Figure 3: Associated Destinations

Table 146, "Associated Destination Fields," on **page 821** shows the fields required for Associated Destinations.

Table 146:   Associated Destination Fields

FIELD NAME	DESCRIPTION	RANGE/DEFAULT
Type	The Associated Destinations types. The Desk, Mobile, and Voice Mail options are auto-created for each User and read-only.	**Range**: None, Desk, Mobile, Softphone, Home IP, Home, or Voice Mail, Desk 2, Softphone 2, Home IP 2, Home 2, Mobile 2 **Default**: None
Description	The description of the Associated Destination (up to 20 characters).	**Range**: N/A **Default**: Blank
Destination (for Desk, Softphone, Home IP, and Voice Mail)	The extension for the Desk, Softphone, Home IP, and Voice Mail associated destination types. This field is a folder link to a programmed device, such as an IP phone or Voice Mail application. These destinations may be an off-node device. If the Main Extension is not provided for a programmed device, "None" is programmed for the User.	**Range**: Any existing extension of permitted device type. **Default**: None
Destination (for Mobile and Home)	The phone number for the Mobile and Home associated destination types. Enter an outside number (up to 48 characters) in the text field.	**Range**: 0 1 2 3 4 5 6 7 8 9 p P * # (p or P is for a pause and set to 3 seconds by default; it is programmable in System\Timers and Limits\Pause Dialing Digit Length) **Default**: Blank
Active (read-only)	Indicates whether the User Destination is to be used. Destinations may be associated with Users but not always active. OAI Applications and UC Advanced can set this flag in conjunction with DND statuses, and so forth.	**Default**: Yes

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Table 146:   Associated Destination Fields (continued)

FIELD NAME	DESCRIPTION	RANGE/DEFAULT
MWI/Pullback	For MWI: Indicates whether or not to propagate MWI from the main extension to the User Destination. See **page 808** for details about MWI Destinations. If you enable MWI for the Mobile destination, DB Programming checks to see if the E-mail Gateway Administrator E-mail Address and E-mail Address fields under E-mail Gateway are blank. If either is blank, the following warning message appears: "Your change cannot be completed because for the E-mail Gateway, either the Administrator E-mail Address or the E-mail Address has not been set up. These may be required by the service provider. Would you like to program the missing setting(s) now?" Click **Yes** to go to the E-mail Gateway folder and make changes, or click **No** to exit without making changes. For Pullback: Indicates whether or not to propagate a Handoff - Pull indication to the User Destination. For example, if you want to pullback calls from you Home IP phone, then you need to set the MWI flag to allow your Home IP phone to know about calls that can be pulled back (i.e., it will light the Handoff key when you are on a Mobile call). If you do not set the MWI flag to Yes, you will not be able to pullback calls to your Home IP from your Mobile phone. Note that to allow your Home IP phone to Handoff - Push a call, then the phone needs to configure the Associated User Extension field (see **page 394**) at the phone level.	**Default**: Yes for Desk and No for Mobile and Voice Mail

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Although the Desk destination for a User is normally the User's Main Extension, special cases may require them to be different, so the Desk and Voice Mail destinations may be changed independently from the Main Extension in the Associated Destinations folder.

NOTE	Further changes to the Main Extension or its Associated Voice Mail will automatically propagate to the Desk destination and Voice Mail destination, so independent programming will be overwritten. The reverse applies as well. If the Voice Mail destination is changed, the Associated Voice Mail of the Main Extension will also be changed.

To change the destination type for a User's associated destination:

1. Select Users – *<User>* – **Associated Destination**.

2. Select the destination type from the Type drop-down list. The available types are None, Desk, Mobile, Softphone, Voice Mail, Home IP, Home, Desk 2, Mobile 2, Softphone 2, Home IP 2, and Home 2. The list displays only the types that are not yet used as a destination type.

To delete a destination:

1. Select Users – *<User>* – **Associated Destinations**.

2. Select the destination that you want to delete.

3. Right-click, and then select **Delete Selected Destination(s)**. If you attempt to delete the default destinations, a warning message appears.

Programming Message Waiting Indication Destinations

For details about MWI destinations, see **page 808**.

To enable MWI for a User's associated destination:

1. Select Users – *<User>* – **Associated Destination**.

2. Select the associated destination type, and then set the MWI flag to **Yes**. The User Destination types that support MWI are Desk, Desk 2, Softphone, Softphone 2, Home IP, Home IP 2, and Mobile.

TROUBLESHOOTING AND TUNING

Redirecting calls to a mobile phone using Dynamic Extension Express typically requires some customer-specific and user-specific configuration and tuning. (See also Table 147 on **page 825**.)

TUNING THE RINGING TIME FOR MOBILE PHONES

Because there is an inherent delay in routing calls out over the public network and the cellular network to a mobile phone, the Ring Duration may need to be adjusted to allow enough "ringing time" before a call is sent on the next step—typically voice mail.
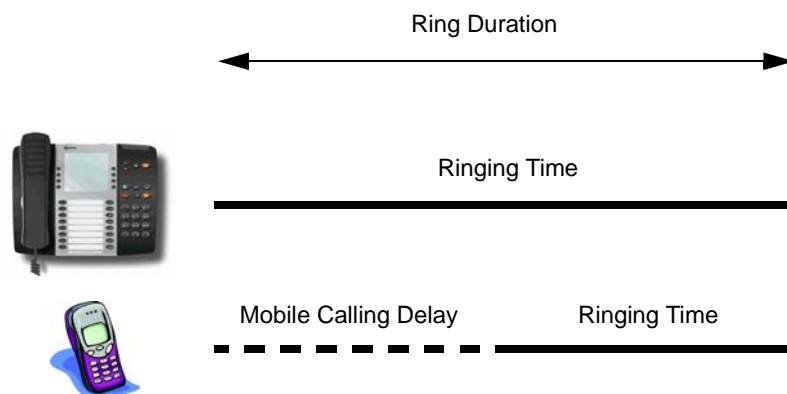
If the ringing time at the mobile phone is too short, the mobile phone user may not have enough time to answer the call before it goes to voice mail. On the other hand, if the ringing time at the mobile phone is too long, the caller may give up before being sent to voice mail.

To tune the amount of ringing time for the mobile phone:

1. Determine the Mobile Calling Delay for a specific user by placing several calls to the user's mobile phone from a 5000 desk phone. Count the number of seconds between the last digit being dialed and the mobile phone ringing. Average the results and consider this to be the Mobile Calling Delay.

2. Ask the user how long (in seconds) he or she wants the call to ring on the mobile phone. This is the Ringing Time.

3. Program the Ring Duration to be the sum of the measured Mobile Calling Delay and the desired Ringing Time.

THE FALLACY OF SIMULTANEOUS RINGING

Although mobile twinning means that the call is offered to both the user's desk phone and mobile phone at the same time, this does not necessarily mean "simultaneous" ringing due to the inherent delay between calling the mobile phone and the mobile phone actually ringing (see the preceding section).

There is no way to eliminate this mobile calling delay, and it varies by carrier and by mobile phone location. Some customers may be tempted to pursue simultaneous ringing by having the first routing step ring only the mobile phone—effectively giving the mobile phone a head start to account for the inherent delay—-before having the second step ring both the desk phone and mobile phone. Not only does this approach not get the mobile phone to ring any sooner, it delays ringing the user's desk phone, which is not the desired outcome if the user is sitting at his or her desk.

So, although some users may expect twinning to mean simultaneous ringing, this is not the case, and it is really only noticeable when testing the feature while sitting at the desk. In typical real-world usage of mobile twinning, users sitting at their desk will hear their desk phone ring and answer it, while users away from their desk will hear their mobile phone ring and answer it. Mitel recommends that customers not pursue simultaneous ringing.

MOBILE TWINNING AND TRUNK USAGE

The mobile twinning capability provided by Dynamic Extension Express can enhance productivity by allowing users to be accessible when away from their main desk. However, mobile twinning effectively consumes an extra trunk for every outgoing call. Trunk capacity may need to be increased to provide for this increased usage. This was the primary reason for providing the Delayed Mobile Twinning template, in which mobile twinning happens only after the call is not answered at the desk phone.

Table 147 provides additional troubleshooting and tuning tips for the Dynamic Extension Express feature.

Table 147: Dynamic Extension Express Troubleshooting And Tuning Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
A call is not following Dynamic Extension Express (DEE).	Do-not-Disturb is overriding Dynamic Extension Express.	If the user does not want DND to prevent DEE from routing calls to other devices, set the "Enable DND Override for Dynamic Extension Express" option to No in DB Programming.
	Manual forward is overriding Dynamic Extension Express.	This is as designed. Manual forwarding overrides DEE.
	Dynamic Extension Express is not enabled.	Perform a user or station dump to see if Dynamic Extension Express is enabled. If it is not, enter feature code **363** on the phone.
	Dynamic Extension Express is not licensed.	Check DB Programming to see if Dynamic Extension Express is licensed. If not, obtain the proper licensing for the feature.
Twinning is not working. The designated mobile or home phone is not ringing.	The mobile number is not programmed correctly, or it is not programmed at all.	Make sure the mobile number is programmed for the user.
	The user's main extension may be toll restricted or does not have outgoing access.	Make sure the user's main extension can place a call to the mobile number.
	The mobile or home number is a long distance number.	Add a toll digit (for example, 1), depending on how ARS is programmed.
DEE users complain that the twinning feature is not ringing the desk phone and the mobile phone at exactly the same time.	There is an inherent delay in routing a call out over the public network and the cellular network.	No action is necessary other than to educate DEE users that both devices are ringing as soon as technically possible, which is what they should want.

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Table 147: Dynamic Extension Express Troubleshooting And Tuning Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
When the user tries the Dynamic Extension Express – Handoff feature (**388**), the display shows NO CALL TO HANDOFF.	The call cannot be handed off.	The user tried to use the Handoff feature on a call that has not been routed by Dynamic Extension Express.
		The user tried to use the Handoff feature on a call that the system has not yet recognized as a valid call.
		The user tried to use the Handoff feature on an unsupported extension. The Handoff feature is supported on the main extension only.
		The user tried to use the Handoff feature on a phantom extension. The Handoff feature is not supported on phantom extensions.
		The user entered the Handoff feature code at the exact time the other party on the call disconnected.
		The call was terminated for some reason.
One or more associated destinations included in a routing step are not ringing.	UC Advanced or an OAI application is setting the Active flag for the specific destination to **No**.	Determine which application is changing the Active flag.
The INVALID ROUTING DESTINATION is displayed on the User's phone.	One or more of the User's associated destinations is not set to Active.	Set the User's associated destinations to Active in DB Programming.
	The User's routing steps have been removed or include one or more undefined destinations.	Make sure the User's routing steps are programmed correctly and include valid destinations.
Callers to DEE users complain of hearing ringing, but not getting an answer.	The Ring Duration might be set longer than a caller would expect before going to voice mail.	Shorten the Ring Duration to the minimum duration that still allows the mobile DEE user enough time to answer the call.
DEE users complain that calls ring on their mobile phones, but they are unable to pick them up before the caller is sent to voice mail.	The inherent delay of routing a DEE call out over the PSTN and mobile carrier network is consuming most of the programmed Ring Duration.	Increase the Ring Duration as needed. Note that increasing the timer too much increases the likelihood of callers giving up instead of waiting for voice mail.
DEE users complain that callers are ending up in the mobile carrier's voice mail application instead of the system voice mail application.	Human Answer Supervision is disabled.	Enable the Human Answer Supervision option.

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Table 147:   Dynamic Extension Express Troubleshooting And Tuning Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
DEE users complain that callers are ending up in the phone system's voice mail application instead of the mobile carrier's voice mail application.	This is normal "as designed" behavior for DEE.	Educate DEE users as to the proper DEE behavior. Or, if desired, disable Human Answer Supervision so that when a mobile carrier voice mail application answers the call, it will be treated like a user answered the call.
DEE users complain that voice mail messages on their mobile phones are simply recordings of the Human Answer Supervision prompt.	The mobile phone's voice mail application is answering the call and recording the Human Answer Supervision prompt.	Shorten the Ring Duration to reduce the likelihood of this happening. Note that shortening the timer too much may not allow the mobile phone user enough time to answer the call.
DEE users complain that they cannot or do not want to press the # key to accept twinned calls.	Human Answer Supervision is enabled, which requires users to press the # key before answering the call.	If desired, disable the Human-Answer-Supervision flag. Note that disabling this feature allows the mobile carrier's voice mail application to answer calls— meaning callers will leave voice mail messages in the mobile carrier's voice mail application rather than the phone system's voice mail application, and no subsequent DEE steps will be processed. If DEE uses trunks without answer supervision (for example, a loop start trunk), set the Enable Human-Answer-Supervision option to Yes, because the call will be immediately answered and routing will be terminated.
DEE users complain that they do not receive caller ID information on twinned calls to their mobile phone.	Outgoing calls to the mobile carrier are going out over trunks that do not support caller ID. PRI and SIP trunks generally support caller ID, but others may not.	Make sure that the proper trunks and system caller ID parameters are set up properly.
DEE users do not want DND on their desk phone to override DEE.	By default, placing the main desk phone in DND mode prevents DEE from routing calls to other devices (for example, mobile or home phones).	Set the "Enable DND Overrides Dynamic Extension Express" option to No. This allows users to put their main desk phone in DND, yet still have DEE attempt to route incoming calls to other devices in the DEE path.
Calls to DEE users stop routing as soon as they advance to a DEE step that includes a mobile phone.	The outgoing call is being routed through a trunk that does not have answer supervision capability (for example, a loop start trunk). Loop start trunks without answer supervision will answer immediately.	Enable the Human-Answer-Supervision flag.

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DIAGNOSTICS

This section details the diagnostics for the Dynamic Extension Express feature. For more information about diagnostics, see "System Diagnostics" on **page 1031**.

DIAGNOSTICS MONITOR

You can use the Diagnostics Monitor utility to access the **Dump Routing Step Templates** function to dump the routing step templates (System Monitor\Dump Routing Step Templates) that are currently defined in the MiVoice Office 250.

INFORMATION DUMPS

You can use the administrator phone and the diagnostics feature code 9900 to dump the following information:

- **Dump Routing Step Templates (RST)**: You can see the routing step templates that are currently defined in the MiVoice Office 250.

- **Dump Extension (9933)**: You can dump the Personal Router Device (PP065 or PRD) information. This provides information about the router device and the calls that are currently being routed.

MID-CALL FEATURES

Mid-Call Features provide a way for mobile users to perform hold, transfer, conferencing and consultation call features when the 5000 places a call via Dynamic Extension Express (DEE). Users who travel often or who rely heavily on a mobile phone can take advantage of the system features.

Mid-Call Features (MCF) are:

- Presented in a user interface with voice prompts, and echo back the user input to avoid any dialing mishaps (it also allows dial-ahead to allow experienced users to skip the prompts).

- Available for Users who are programmed to use the Dynamic Extension Express (DEE) feature (which is controlled by a flag in User programming and is disabled by default).

FEATURE ACTIVATION

MCF uses two feature codes to control the voice-guided feature system. The codes are programmable by the system administrator and can be from 1 to 2 digits in length (0-9, *). See"User Programming Fields" on **page 788** for detailed information.

> **NOTE** All Activation and Confirmation codes will be heard by the other party. Any digits entered while in the feature menus will not be heard by the other party.

While on an external system call initiated from DEE, the user may enter the MCF activation code. Then the MCF main menu will begin playing. At this point, the original call is now on hold and will hear hold audio based on the trunk group database setting for holding calls (this is a new database field primarily for MCF; see "Audio for Calls Holding for this Device" on **page 517**).

> ⚠️ **CAUTION: It is possible for digits pressed by the other party to activate MCF on the local party due to large amounts of echo when using Media Stream Resources (MSR) DSP-based audio resources used for DTMF detection and audio playback. While this scenario is really not realistic, it is possible and steps should be taken to eliminate as much echo as possible on loop-start and T1 central office connections. Systems using Dual T1 cards do not have this issue since they use built-in DTMF receivers and not MSRs.**

MCF DTMF DETECTION ANALYZER

The system employs a simple algorithm to prevent accidentally activating MCF in situations were DTMF is required by the far end (i.e. voicemail, pin code, etc).

If the activation/confirmation code is only one digit, the system makes sure that no other digits have been pressed within a set timeframe. This allows the user to enter in a string of digits that may contain the activation code without actually activating MCF. If there are two digits in the code, the second digit must be entered within three seconds of the first.

MCF PROMPTS

MCF has voice-guided prompts for all supported features. Once MCF is activated, the caller will hear the context-sensitive options based on current features that may already be in progress (such as an active conference).

- The user is given all of the features that are currently supported along with appropriate DTMF digit(s) to initiate them. Many of the features require extra inputs such as Conference and Transfer.

- Any digits pressed while the prompts are playing will be processed immediately which allows the user to use dial-ahead to skip the voice prompts.

- The user can return to the menu one level above the current menu by pressing asterisk (*).

- Feature menus will time out to the previous menu after 3 cycles if no input is received. If the main menu times out, the user will automatically be reconnected to the holding call.

- The MCF prompts will be played in the language programmed at the user's main extension. If the main extension language is not supported on the current node, the current node's primary language will be used.

- After the MCF digits are entered, the system plays a confirmation prompt that echoes back the user input. The user can confirm the input or re-enter the input if incorrect.

MCF FEATURES

The following table shows all features available to the user.

Table 148:   MCF Features

FEATURE	DTMF	EXTRA INPUT 1	EXTRA INPUT 2
Re-Access Holding Call	1		
Transfer	2	Transfer Destination Type	Extension/Outside Number
Conference	3	Conference Party Type	Extension/Outside Number
Make Consultation Call	4	Call Destination Type	Extension/Outside Number

The user can also use these codes while using MCF:

- Return to the previous menu: * (default value)

- Activate or Cancel: *# (default value)

- End Call: # # (default value)

If a feature call (such as transfer call, conference call, or consultation call) is disconnected from the far end, the user will return to the main menu

Re-Access Holding Call

Once mid-call features are activated, the original call is put on hold. At any time, the user may re-access the holding call by entering option 1 from the main menu.

Transfer Call

The user may enter the transfer call menu by entering option 2 from the main menu. Both blind and announced transfers are supported. Once in the transfer call menu, the user is provided with the following options for the transfer destination:

Table 149:   Transfer Call Menu Prompts

TRANSFER DESTINATION	DTMF	DESCRIPTION
Internal	1	Transfer the call to an internal device using the extension.
External	2	Transfer the call to the public network using an outside number.
Operator	3	Transfer the call to the operator.
Auto Attendant	4	Transfer the call to the auto attendant to lookup an extension.
Voicemail	5	Transfer the call to your voice mailbox.

If the user does not have a voice mailbox or an auto attendant application does not exist on the system, those options will not be available from the menu.

The user may complete the transfer either by entering in the MCF confirmation code (##) or by hanging up the call.

Conference Call

The user may enter the conference call menu by entering option 3 from the main menu and then select from the following options for the new conference party:

Table 150: Conference Call Menu Prompts

NEW CONFERENCE PARTY	DTMF	DESCRIPTION
Internal	1	Add new internal conference party device using the extension.
External	2	Add new external conference party using an outside number.
Operator	3	Access the operator to lookup party to add to conference.
Auto Attendant	4	Access the auto attendant to lookup party to add to conference.

Once the called party has answered, the user may add them into the conference by using the MCF confirmation code (**# #**).

If a conference did not previously exist, it will be created and all parties will be connected (including the original holding call).

If all conference resources are exhausted, an error tone sounds and the user will be reconnected to the conference party call. The user may try to complete the conference at a later time if desired.

Consultation Call

A consultation call allows the user to place the current call (or conference) on hold, make a second call, and then return to the holding call when the consultation call is over. The user may enter the consultation call menu by entering option 4 from the main menu and then select from the following options for the consultation call destination:

Table 151:

CONSULTATION CALL DESTINATION	DTMF	DESCRIPTION
Internal	1	Call to an internal device using the extension.
External	2	Call to the public network using an outside number.
Operator	3	Call to the operator.
Auto Attendant	4	Call to the auto attendant to lookup an extension.
Voicemail	5	Call to your voice mailbox to listen to your messages.

NOTE	A consultation call may not be converted to a transfer announcement or conference call.

MCF PROGRAMMING

See the *MiVoice Office 250 DB Programming Help* for detaiils. Also, see the following sections:

- "Audio for Calls Holding for this Device" on **page 517**

- "User Programming Fields" on **page 788**
- "User-Related Information" on **page 806**

MCF INTERACTION WITH OTHER FEATURES

The following features in the system interact with MCF:

- Hold Recall: The original call on hold will never recall. This allows the user ample time to perform any features they want to perform without worrying about the holding call recalling.

- Hold: A Mid Call Feature call may be placed on hold by the other party. While the MCF call is on hold, the user may not perform any other MCF features. However, the user may cancel the current MCF call at any time and return to the main menu.

- Transfer: Since the MCF call can also be transferred to a new destination. This allows an Auto-Attendant or IVR application to transfer the call to another device within the system.

- Reverse Transfer: Calls put on hold for a mid-call feature cannot be reverse transferred. This includes the original holding call and any calls related to an ongoing MCF call.

- Emergency Dialing:
  - Internal Call Option: If the user selects the internal call option for consultation call, transfer call, or conference call, and they dial an emergency feature code, the call will fail and the user will receive an error indication.
  - External Call Option: If the user selects the external call option for consultation call, transfer call, or conference call and they dial an emergency number, the call will be placed as dialed. This is NOT recommended since the system will be placing the call and the location will be the system, not the external device

- Peer-to-Peer Media: The system disables Peer-to-Peer Media for calls to users using MCF.

- DEE Handoff and Pullback: A call cannot be handed off or pulled while an MCF call is in progress.

- Caller ID: It is recommended that the main extension's Calling Party Name and Calling Party Number be programmed properly so that, when using MCF, the caller ID information will correctly identify the person calling.
  - Internal Calls: The internal destination will see that the call is from the user's main extension if the database fields mentioned above are filled in. If not, the internal destination will see "TRNK GROUP X RINGING IN".
  - External Calls: All external calls made by MCF will send caller ID based on the main extension database programming. If the trunk group used to make the external call has the Forced Trunk Group Calling Party Name and Number flag set to Yes, the external party will see the Trunk Group Calling Party Name and Number rather than the main extension Calling Party Name and Number.

MULTI-NODE SUPPORT

Mid Call Features are supported in a multi-node environment; however both the node where the User exists and the node where the call will interface with the public network need to be running software release 6.0 SP1 or greater.

ADMINISTRATOR-RELATED INFORMATION

This is located in Users\<*User*>*Administrator-Related Information*.

Maintenance Accounts (known as System Accounts prior to Database Programming Release 6.0) are now associated with users. Programming for users who are Administrators is performed in this folder. The following items are programmed:

- **Enable Advanced Administrator Access**: Indicates if the user is an advanced system administrator. Default = No.
  - Intended to be used by system administrators and field technicians.
  - The Customer Configurator user type will have access to the User Web Portal if there is a main extension. Otherwise they will have administrator web access only (for adds, moves, and changes).This user type has access to all menu items and folders in DB Programming.
  - The Administrator account user can edit the Customer Configurator account password in the Maintenance Accounts folder, and also can add or remove other accounts, enable/disable the support account, edit descriptions and access rights, and edit other account passwords.
  - The Administrator account is the only system account enabled in a default database.
  - The default password for both the Customer Configurator and Administrator accounts is set to itpassw.
- **Enable E-mail Alarm Notification**: Indicates if the user will receive alarm notifications from Sys Ad via e-mail. Default = No.
- **Enable Customer Configuration Access**: Indicates if the user has Customer Configurator privileges in Database Programming. Default = Yes.
- **Enable Advanced Web Page Programming and Diagnostics Access**: Indicates if the user has access to the Administrative Web Portal. Default = No.

Chapter 14

VOICE PROCESSOR FEATURES AND
PROGRAMMING

INTRODUCTION

This chapter describes voice processing system features and programming for the MiVoice Office 250. For mailbox programming, see "Subscriber Mailboxes" on **page 959**.

For voice processing system installation information, refer to the applicable product installation documentation as described in "Mitel Voice Processing Systems" below.

For voice processing-related troubleshooting issues, see "Troubleshooting a Voice Processing" on **page 1018**.

PROGRAM PLANNING SHEETS

For program planning sheets that list the voice processor system-wide information, refer to the programming planning sheets provided on the software DVD-ROM. Program planning sheets are also available on the Mitel eDocs Web site (http://edocs.mitel.com).

MITEL VOICE PROCESSING SYSTEMS

NOTICE
Voice Processing Unit (VPU) end of sale. VPU is no longer supported in v3.0 or later. The VPU was discontinued in May 2007 and has reached its end of sale. Mitel recommends that current VPU installations upgrade to either MiCollab Unified Voice Messaging (UVM) or MiCollab Unified Messaging. You cannot convert a VPU database to an MiCollab Unified Messaging database. Contact your local provider for more information.

The MiVoice Office 250 supports the following voice processing applications:

- **MiCollab Unified Voice Messaging (UVM)**: The preinstalled internal voice mail application that includes Unified Voice Messaging functionality. You program MiCollab Unified Voice Messaging options entirely in MiVoice Office 250 Database (DB) Programming. For more information about MiCollab UVM, see **page 838**.

- **MiCollab Unified Messaging**: An external voice processing system that resides on the Mitel Application Suite® (MAS) server and uses Session Initiation Protocol (SIP) to communicate with the MiVoice Office 250. MiVoice Office 250s support MiCollab Unified Messaging as the system voice processing application. MiCollab Unified Messaging is installed as a separate, external voice mail processor. For more information, refer to the following resources:

  - *MiVoice Office 250 and MiCollab Unified Messaging Integration Guide*
  - MiCollab Unified Messaging System Administration Help
  - MiCollab Unified Messaging Technician's Handbook
  - *MiVoice Office 250 Installation Manual*
  - MiVoice Office 250 DB Programming Help

SUPPORTING DOCUMENTATION

Because various voice mail products work with the MiVoice Office 250, this section does not include voice mail subscriber (user) feature instructions. Refer to the MiVoice Office 250 Unified Voice Messasing, and Embedded Voice Mail Card Voice Mail User Guide for your system for feature descriptions and instructions.

Refer to the following books for more information about voice processing features:

- *Unified Voice Messaging, and Embedded Voice Mail Card User Guide,* part number 835.3205: Provides voice mail system feature descriptions and end-user instructions.

- *MiVoice Office 250 Phone Administrator Guide,*: Provides feature descriptions and instructions for administrator phone administrator features.

- *MiVoice Office 250 Voice Mail Administrator Guide,*: Provides feature descriptions and instructions for administrator voice mail administrator mailbox features.

- *MiVoice Office 250 Installation Manual :* Provides MiVoice Office 250 installation, specification, and maintenance information.

- *MiVoice Office 250 System Administration & Diagnostics Guide,* and *Mitel System Administration & Diagnostics Help:* Provide 5000 System Administration & Diagnostics installation, specification, maintenance, and diagnostics information.

For voice processing specifications and capacities, refer to the "Specifications" chapter in the *MiVoice Office 250 Installation Manual* .

UVM VOICE PROCESSING SYSTEM

This section contains the following information:

- **Feature Description** below

- Disabling UVM for External Voice Processing on **page 841**

- Troubleshooting Unified Voice Messaging on **page 842**

FEATURE DESCRIPTION

Built-in on the MiVoice Office 250, UVM runs on the Linux® operating system and provides voice messaging services. On systems not equipped with a PS-1, language prompts, UVM messages, system applications, and the customer database are stored on a compact flash-type memory card that resides in the Base Server. On systems equipped with a PS-1, the PS-1 supports the UVM application, which relieves the Mitel memory card of voice messaging tasks.

The size of the compact flash-type memory card installed in the system determines the capacity of UVM message storage, which ranges from approximately 1800 minutes with a 256 MB card to more than 12,000 minutes with a 1 GB card. For more details, see "UVM Voice Message Storage Capacities" on **page 839**.

In a network scenario, MiVoice Office 250 nodes equipped only with UVM cannot forward messages to, or receive messages forwarded from, mailboxes on other nodes. For UVM storage capacities, see "UVM Voice Message Storage Capacities" on **page 839**.

Only applicable to SMTP–MIME e-mail servers, the Forward to E-Mail feature allows users to send voice mail messages as **.wav** file attachments. The feature is configured through DB Programming, and the voice mail server is responsible for security issues.

REFERENCES FOR MANAGING UVM AND MITEL MEMORY CARDS

The following are references for managing UVM and Mitel memory card issues:

- For the applicable procedure required to upgrade a memory card, refer to *MiVoice Office 250 Memory Card Replacement Instructions*, part no. 835.3033.

- For UVM troubleshooting guidance, refer to the *Voice Processing Diagnostics Manual*, part no. 550.8019.

- The MiVoice Office 250 also supports voice mail on optional external voice mail systems such as MiCollab Unified Messaging (see "Mitel Voice Processing Systems" on **page 836**).

- For UVM troubleshooting information, refer to the *Voice Processing Diagnostics Manual*, part no. 550.8019.

UVM PORT LICENSING AND IP RESOURCE ALLOCATION

IMPORTANT	Because UVM ports are licensable, customers upgrading their port capacity are required to purchase the corresponding licenses.

With v5.0 or later, UVM can be licensed for up to 24 or 32 voice mail ports. The original CS Controller Base Server supports up to 24 ports. The HX Base Server and PS-1 equipped

platforms support up to 32 ports.The ports are allocated dynamically up to the number of valid UVM licenses. Ports are licensed four at a time and may *not* be licensed individually.

UVM can be expanded from 4 ports to 16 ports, which provides increased availability and response time for voice processor applications. UVM runs on the Processor Module (PM-1) processor the same way as with previous versions, but more ports are available.

The number of UVM ports available contributes to fast response to voice processing applications such as Voice Mail, Message Notification/Retrieval, Automated Attendant, and Record-A-Call. When sufficient Voice Mail ports are not available, users waiting for a port are camped-on, or they are sent to Music-On-Hold or down some other path. Having more UVM ports makes the system operate optimally.

The number of UVM ports is configured through DB Programming. When the number of ports has been changed, the system must be reset to reallocate the IP resources shared between the IPRA and UVM.

In offline mode, the user is prompted for the voice mail type so the proper logic can be followed.

UVM VOICE MESSAGE STORAGE CAPACITIES

> **NOTICE**
>
> MiVoice Office 250 v3.0 or later requires a minimum 512MB compact flash-type memory card. If this system is currently using a 256MB compact flash-type memory card, upgrade the card to either 512MB (part number 841.0274) or 1024MB (part number 841.0273).

The amount of message storage, which is allocated for each user as a quota of the total storage capacity, is programmed in DB Programming. A warning flag can be set for each mailbox to alert the user when a specified percentage of the allocated quota has been reached. For more information, see "Subscriber Mailboxes" on **page 959**.

The UVM Forward to E-Mail feature requires significant memory for converting voice mail messages to the image file format. A system not equipped with a PS-1 requires a minimum 512 MB memory card that use the feature.

In addition to the standard American English prompts, each set of language prompts loaded on a system reduces the amount of storage available by approximately 1.25 hours (75 min.). For the most part, the reduction of memory capacity due to adding language prompts impacts system not equipped with a PS-1. A system equipped with a PS-1 relies on hard disk storage rather than the system memory card, which frees up memory for other system tasks. The MiVoice Office 250 also supports British English, Spanish, Canadian French, and Japanese language prompts.

NOTE The Japanese language is not supported on 52xx/53xx-series IP phones. For complete information about multilingual capability, see "Multilingual Capability" on **page 314**.

A system not equipped with a PS-1 can use the G.726-32 or G.729 codecs for voice compression.

G.726-32 uses 32 kbps; G.729 compression uses 8 kbps. The default UVM compression is set to G.729 to maximize voice mail storage capacity. Using G.726-32 consumes four times as much storage resulting in 25% of the storage capacity, but it also produces better audio quality

for voice messages in some scenarios. There are some scenarios in which G.726-32 may be appropriate for UVM compression:

- In some environments where G.729 is used for IP phones, multiple compression/decompression steps may result in degraded voice quality of the stored messages. Using G.726-32 for voice mail storage may improve the quality of recorded voice.

- The VPIM feature, as specified in the VPIM v2 standard, natively uses G.726-32 compression for the interchange between voice mail systems. By using G.726-32, the MiVoice Office 250 does not need to transcode from G.729 to G.726-32. The result is more efficient processing.

UVM on a system equipped with a PS-1 does not offer a vocoder selection because it stores voice messages on the PS-1 hard disk. Because storage is plentiful, no compression is used (G.711). If the resulting voice storage capacity is unacceptable, upgrade to a larger compact flash-type card. Refer to the *MiVoice Office 250 Installation Manual* for UVM storage capacities.

UVM ON A SYSTEM EQUIPPED WITH A PS-1

On a system equipped with a PS-1, UVM supports up to 32 voice mail ports. UVM is hosted on the PS-1. UVM on the PS-1 can be configured with 0, 4, 8, 12, 16, or 32 ports in accordance with feature licensing.

Table 152 shows the relationship between the number of valid UVM licenses and the maximum number of active UVM ports in a system equipped with a PS-1. DB Programming does *not* allow dynamic allocation of more UVM ports than are licensed for the system and that have been manually programmed into the appropriate Time Slot Group.

Table 152: UVM Licenses, Time Slot Group Ports, and IP Resources Allocated

VALID UVM MAIL PORT LICENSES	PORTS DEFINED IN TIME SLOT GROUP	IP RESOURCES DYNAMICALLY ALLOCATED	ACTIVE UVM VOICE MAIL PORTS
0	0	0	
4	4	0–4	
8 (4 + 4)	8	0–8	
12 (4 + 4 + 4)	12	0–12	
16 (4 + 4 + 4 + 4)	16	0–16	
32 (4 + 4 + 4 + 4 + 4 + 4 + 4 + 4)		0–32	

If all of the IP resources available for UVM are in use, the next caller is camped-on to wait for a resource to become available.

UVM FORWARD TO E-MAIL

The UVM Forward to E-Mail feature provides a single retry rule that applies to all domains and all errors. It specifies a retry every 15 minutes for one hour. After one hour the e-mail message is deleted from the system.

Retries are made if the software fails to send the e-mail message to the SMTP server, which may occur if the SMTP Server Address is configured incorrectly. The voice mail message is unaffected by the retry logic. If the e-mail address is configured incorrectly at the voice mailbox, but the software can send the e-mail to the SMTP server, then the message is *not* queued for retry.

DISABLING UVM FOR EXTERNAL VOICE PROCESSING

If you are installing an external voice mail system, you must disable UVM *before* you install and program the external voice processing system.

NOTE	After disabling UVM, System Alarm 203 appears on both the LCD panel on the Base Server and on the administrator phone (normal functionality). This alarm appears when a voice mail application is disconnected from a MiVoice Office 250 that previously had a voice mail system connected.
	The Disable Unified Voice Messaging menu option is not dimmed after disabling Unified Voice Messaging. This allows you to disable UVM if you did not do so before installing and programming the external voice processing system. If necessary, use the Administration Web Portal (AWP) to verify that UVM is disabled.

To disable Unified Voice Messaging:

1. Open a DB Programming session.

2. Back up the Voice Mail database. You must use a USB flash drive for this operation. See "Saving and Restoring Voice Processing Databases" on **page 1001**.

3. From the Mitel DB Programming menu bar, select Operations – Voice Processor Operations – **Disable Unified Voice Messaging**.
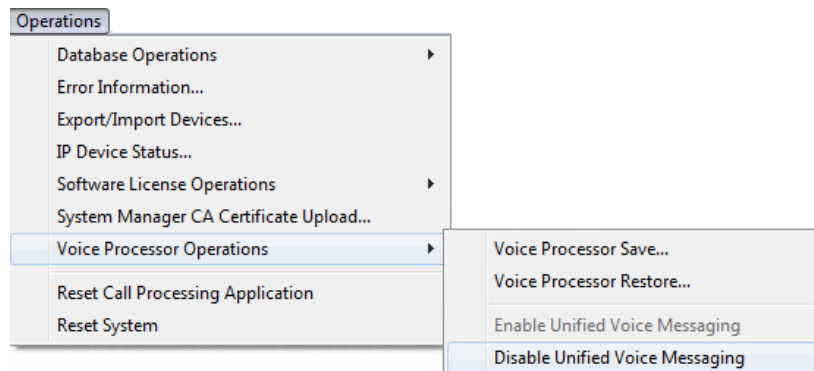
4. Click **OK** at the prompt to disable Unified Voice Messaging and terminate the session.

5. Restart the session to configure external voice processing parameters, if needed.

TROUBLESHOOTING UNIFIED VOICE MESSAGING

Table 153 summarizes the troubleshooting strategies recommended for resolving discrepancies that may occur with Unified Voice Messaging (UVM).

Table 153:   Unified Voice Messaging Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE
UVM is running on the PS-1. IP phones or digital telephones have bad quality.	Voice packets are not getting from the Base Server to the PS-1 in a timely manner. This could happen for the following reasons: • Network is overloaded • PS-1 and Base Server are not on the same LAN and VLAN using the same switch
UVM is running on the PS-1. IP phones or digital telephones have bad quality.	Voice packets are not getting from the PS-1 to the phones in a timely manner. This could happen if PS-1 or any of the IP phones have duplicate IP addresses. Power cycling can sometimes cause routers/switches to default to previous values.

INTERACTION BETWEEN SYSTEMS

When using an external voice processing system, the software and hardware that control the Voice Processing applications is located in the external voice processing system.

The MiVoice Office 250 must be able to communicate with external voice processing systems for the applications on those systems to function. If the system is unable to communicate with external Voice Processing systems, the applications cannot be used and the following occurs:

- Direct ring-in calls to an application are sent to the primary attendant (if there is one).

- Intercom calls to applications receive reorder tone. Display phones show <Application name> IS UNPLUGGED.

- Administrator phone users see an Alarm 203 (Voice Processing: Communications Link Down) if the system detects a loss of communications between the phone system and the external voice processing system.

- Calls do not recall to an application. Instead a recalling call remains at the phone or hunt group and ring until it is answered or the Abandoned Call timer expires.

- If an application is used as a hunt group overflow or announcement phone, calls are not sent to the application, but remain camped on to the hunt group.

- A transfer to an application camps on until the Recall timer expires. Then it follows the usual recall path.

- Calls cannot be forwarded or system forwarded to an application.

- SMDR information is not recorded in the buffer on the external voice processing computer hard drive.

- Database programming for external voice processing features is not allowed.

VOICE PROCESSING SYSTEM NETWORKING

The voice processing system network is programmed separately from the phone system network. Up to 100 nodes can exist on the voice processing system network. The following programming fields provide networking support:

- **Node**: The Network Node table contains information about each Voice Mail node in the network. This table contains information that allows the system to identify nodes, contact nodes, and deliver messages to nodes. Each node has its own internal message queue (similar to mailbox message queues) and stores messages destined for other nodes until they are delivered. See **page 847** for programming information.

- **Network Mailbox**: A network mailbox is a mailbox on the local voice processing system that accesses a mailbox on a remote node or the node itself. A Network Mailbox is very similar to an Extension ID. If you log on to a Network Mailbox, the only options available are the directory name, the password, and the transfer method. There are two types of network mailboxes, a "direct network mailbox" and a "node network mailbox."

    - A *direct network* mailbox is a network mailbox that accesses a mailbox on a remote node. To do this, the network mailbox is programmed to use a specific mailbox number.

    - A *node network* mailbox does not refer to a specific mailbox on a remote node, but instead, refers to a remote node in general. To do this, the mailbox is not programmed with a specific mailbox number. This tells Voice Processing to send the call to the remote node programmed. The directory name for this type of mailbox should be something that indicates the location of the node rather than an individual's name. For example, "Denver" or "Main Office."

Each node is programmed with a Network Type field that identifies how the local node communicates with the remote node. Remote voice processing nodes can use the following protocols:

- VPIM Networking on **page 844**

- on **page 845**

For information about network operation overview, see **page 845**.

VPIM NETWORKING

Voice Profile for Internet Mail (VPIM) protocol communicates with other voice processing systems using the VPIM protocol. VPIM networking requires the Voice Processor Messaging Networking software license. See **page 851** for details about VPIM Networking.

NOTE	MiCollab Unified Messaging systems must be configured to use the G.721 codec when using VPIM to communicate with MiCollab UVM systems.

The VPIM service is enabled on the originating and destination voice processing systems. Messages are exchanged through e-mails sent between the voice processors. Each message is an attachment to each e-mail being exchanged between the two voice processors. See the following example.
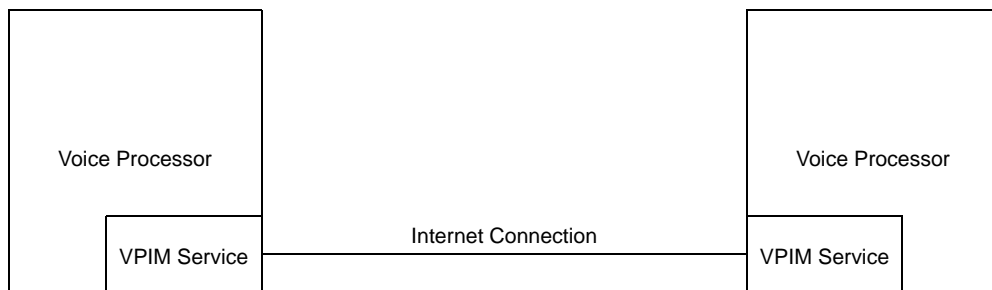
NETWORK OPERATION OVERVIEW

The Network Node table is used to define information about each node in the Voice Mail network. Before the local node can send messages to a remote node, there must be an entry in the Network Node table for the remote node.

To leave messages for mailboxes on remote nodes, you must have "network mailboxes" on the local node. If the local node only has network mailboxes associated with specific mailbox numbers, the local node can only leave messages for those mailboxes. If the local node has a network mailbox for a remote node (no mailbox number specified), the local node can leave messages for any mailbox on the remote node, if the "Validate Off-Node Mailboxes" option is disabled.

LEAVING MESSAGES

Users can leave messages for any network mailbox as either a subscriber or a non-subscriber. Leaving messages for a direct network mailbox is the same as leaving messages for a regular mailbox. After the user enters the network mailbox number, the Voice Mail application announces the mailbox directory name, if it is recorded, or the mailbox number followed by the recording instructions. Following the CRA instructions, the caller can leave a message.

Users can also leave messages using a node network mailbox. Users who call the node network mailbox hear a prompt asking them to enter the desired mailbox on the remote Voice Processing system. After that step, it is the same as leaving a message for a regular mailbox. Node network mailboxes are useful when you do not know the direct network mailbox number and/or when the mailbox on the remote node does not have a direct network mailbox on the local node. (If Voice Processing attempts to deliver the message from a local mailbox user to the remote node and the mailbox number is invalid, the message is returned to the sender.)

If the Validate Network Mailboxes flag is enabled, users can only leave messages for mailboxes on remote nodes that have a direct network mailbox on the local node. If there are no direct network mailboxes that refer to mailboxes on the remote node, the caller hears a prompt indicating that the message cannot be delivered to the remote node.

Unlike regular mailboxes, network mailboxes (both direct and node mailboxes) do not have their own message queues. When a message is delivered to a network mailbox, Voice Processing looks up the mailbox node information, and stores the message locally. The messages are stored until they are transmitted to the remote node or returned to the sender.

UNDELIVERABLE MESSAGES

Voice Processing delivers messages to another node by calling the remote node System Number. If the remote number is busy or there is no answer, Voice Processing continues to attempt to contact the remote node until the remote node answers, or until it has reached the Maximum Network Call Attempts limit. If Voice Processing has made consecutive, unsuccessful Maximum Network Call Attempts (due to busy or no answer), it stops attempting to contact the remote node and return any messages pending for the node. The messages are then considered "undeliverable." Undeliverable messages pending for a node are handled according to the Undeliverable Network Messages Destination field:

- When the field is set to *Delete*, all undeliverable messages are deleted.

- When the field is set to *Sender*, all undeliverable messages are returned to the sender's mailbox, if possible. If the sender is unknown, the messages are returned to the System Administrator's mailbox. If the System Administrator's mailbox does not exist, then the messages are deleted. When a subscriber listens to a returned message, he hears a prompt indicating that the message was undeliverable.

- When the field is set to *System Administrator*, all undeliverable messages are returned to the System Administrator's mailbox, if it exists. Otherwise the messages are deleted.

VOICE PROCESSOR NODES

The Node programming area is used to identify and define information about each node in the voice processor network. Before the local voice processor can deliver messages to another voice processor node in the network, the remote node must have an entry in this table. See **page 844** for a complete description of voice processor networking.

Each node has its own internal message queue, which is similar to mailbox message queues, and will store messages destined for other nodes until they are delivered. Up to 100 nodes can exist in the voice processor network.

LOCAL NODES

Each MiVoice Office 250 has a local node for handling system and subscriber voice processing features. The local node can be a UVM or MiCollab Unified Messaging voice processor.

UVM voice processing systems can use Voice Profile for Internet Mail (VPIM) protocol to connect to remote node or third-party voice processing applications. For more information, see "Voice Profile for Internet Mail (VPIM) Networking" on **page 851**.

To view the local voice processing node:

Select Voice Processor – Devices – **Nodes**.

CREATING NODES

To create a node:

1. Right-click in the right side of the screen and select **Create Node**. DB Programming will need to close to reset. When you restart the session, the new node appears in the Nodes folder.

2. Program the following fields for each node:
   - **Number**: The network node number is a number between PP200 and PP299. The number is assigned automatically when the node is created and cannot be changed.
   - **Description**: The Description field is used to provide a meaningful name for the node (such as, "Phoenix" or "Houston"). This field can contain up to 20 characters. The default for this field is "Node xxx." For example, node 1 defaults to "Node 001" and node 10 defaults to "Node 010."
      a. Enter the desired description in the text box.
      b. Press **ENTER** or click another field to save the change.
   - **System Number**: The System Number field is either a phone number or an IP address.
      a. Enter the appropriate number in the text box.
      b. Press **ENTER** or click another field to save the change. If the Network Type field is set to:
      - *CO AMIS*, the System Number field is a number used to connect to the remote

node. This number should ring directly into the remote Voice Processor.

- *TCPIP*, the System Number field should be an IP address.

- *IC AMIS*, the System Number field is **not** programmable.

• **Extension**: The Extension field is available only when the Network Type field is IC AMIS.

   a. Right-click the field and select **Change Extension**.

   b. Program the phone, CO trunk group, or hunt group extension.

• **Network Type**: The Network Type field identifies how the local node communicated with the remote node. This field can be programmed to None, IC AMIS, CO AMIS, or TCPIP.

   a. Select the drop-down list box, then scroll to the desired setting.

   b. Press **ENTER** or click another field to save the change.

   - If this field is programmed to None, the node will be unavailable for use.

   - AMIS is an analog networking protocol that uses the public switched telephone network (PSTN). UVM does not support AMIS networking.

   - TCPIP is a direct network connection between the local Voice Processor and the remote Voice Processor.

REMOTE NODES

The following sections describe remote nodes and options for voice processing systems.

CREATING A REMOTE NODE

You can create a remote voice processing node for UVM. UVM systems use the VPIM protocol to communicate with the local voice processor. See "Voice Profile for Internet Mail (VPIM) Networking" on **page 851** for more information.

To create a remote node:

1. Select Voice Processor – Devices – **Nodes**.

2. Right-click in the right pane, and then click **Create Node**. The new node appears in the node list.

PROGRAMMING REMOTE NODE OPTIONS

When the voice processor network type is set to use TCPIP, you can program the following node options.

Start and Stop Times

These options specify the time of day that messages are delivered to the remote node. If the start time and stop time are the same, it indicates that deliveries are valid the entire day. The default value for this field is 8:00AM to 8:00AM.

To set the time:

1. Select Voice Processor – Devices – **Nodes**.

2. In the **Value** column, select the current Start or Stop Time.

3. Click a time field to change the setting, as shown in the following example. For example, if you want to change the hour, click the hour, and then select the hour from the list. Repeat for the Minutes and AM/PM fields.

Hour is selected
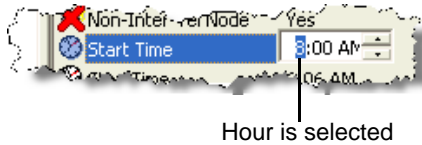
4. Click out of the field or press **ENTER** to save the change.

Days of the Week

These options specify the days of the week that messages are delivered to the remote node. Any combination of the days of the week is valid.

To enable a specific day of the week:

1. Select Voice Processor – Devices – **Nodes**.

2. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

3. Click out of the field or press **ENTER** to save the change.

Remote Node Timers and Limits

You can program either of the following timers and limits for each node.

*   **Latency Time**: This specifies how long the Voice Processor will wait between successful outgoing calls to the remote node if there are pending messages for the remote node. This field is only in effect if the current time on the Voice Processor is within the remote node delivery time/date settings. The range for this field is 0–1440 minutes (24 hours). When this field is programmed to 0, it indicates that the Voice Processor will deliver messages immediately to the remote node. In other words, as soon as the local node receives a message that is destined for the remote node, the Voice Processor will attempt to connect to the remote node and deliver the message (provided that the time is within the remote node delivery time/date settings). If this field is something other than 0, it specifies the amount of time that a message will remain pending on the local node before the Voice Processor attempts to deliver it to the remote node. For example, if this field is programmed to 60 minutes, the next network call to the remote node will occur 60 minutes after the last successful call to the node. The default value for this field is 30 minutes.

*   **Priority Latency Time**: This specifies how long the Voice Processor will wait between outgoing calls to the remote node if there are pending priority messages for the node. This field is only in effect if the current time on the Voice Processor is within the remote node delivery start and stop date/time settings. The range for this field is 0–1440 minutes (24 hours). When this field is programmed to 0, it indicates that the Voice Processor will deliver

priority messages immediately to the remote node. Any time the Voice Processor delivers pending priority messages, it will also deliver any non-priority messages that are pending for the remote node. Note that this field takes precedence over the Latency Time field. The default value for this field is 5 minutes.

- **Message Threshold**: This specifies the number of messages that must exist in the node message queue to force the Voice Processor to place a call to the remote node. This field is only in effect if the current time on the Voice Processor is within the remote nodes delivery time/date settings. This field can be used to ensure that the local Voice Processor does not get too backed up with messages destined for the remote node. If this field is programmed to 1, every message the local node receives that is destined for the remote node is delivered as soon as it is received by the local node. If this field is programmed to another number, it means that as soon as the remote node message queue has that many messages pending, the Voice Processor will attempt to connect to the remote node and deliver the messages. Note that this field takes precedence over both the Latency Time and Priority Latency Time fields. The range for this field is 0–100 messages and the default value for this is 5 messages.

To program remote node timers and limits:

1. Select Voice Processor – Devices – **Nodes**.

2. Double-click **Timers and Limits** to view the list.

3. In the **Value** column, click the value, and then type the new information in the box.

4. Click out of the field or press **ENTER** to save the change.

VOICE PROFILE FOR INTERNET MAIL (VPIM) NETWORKING

This section contains the following information:

- **VPIM Licensing** below

- **VPIM Timers and Limits** below

- VPIM Messages on **page 852**

- Overview of VPIM Programming on **page 852**

- Troubleshooting VPIM Networking on **page 857**

NOTE	MiCollab Unified Messaging systems must be configured to use the G.721 codec when using VPIM to communicate with UVM systems using G729. G.721 on MiCollab Unified Messaging is compatible with G.729.
	This section details VPIM for UVM only. VPIM for MiCollab Unified Messaging is programmed through the MiCollab Unified Messaging Web Console. For details about VPIM with MiCollab Unified Messaging, refer to the *MiCollab Unified Messaging System Administration Help* and the *MiCollab Unified Messaging Technician's Handbook.*

Voice Profile for Internet Mail (VPIM) is a method for encoding voice mail messages as data, enabling travel via Simple Mail Transfer Protocol (SMTP) over IP networks. When VPIM is configured and enabled on two VPIM-compliant voice processors, the systems can exchange voice messages, sent as e-mail attachments, over the Internet.

VPIM uses the fully qualified domain name (FQDN) for sending and receiving VPIM messages. The FQDN comprises the hostname and the domain name. For example, "myserver.mycompany.com," where "myserver" is the hostname and "mycompany.com" is the domain name.

VPIM networking on the MiVoice Office 250 adheres to the VPIM v2 standard as described by the Internet EngineeringTask Force (IETF) Request for Comments (RFC) 2421, http://www.ietf.org/html.charters/OLD/vpim-charter.html.

VPIM LICENSING

This feature requires a "Voice Processor Messaging Networking" software license.

VPIM TIMERS AND LIMITS

The Maximum Network Message Length limit governs the VPIM message size. See "Network Settings" on **page 859** for more information about the Maximum Network Message Length limit. Also, the E-Mail Retrieval Interval (Minutes), for UVM systems only, indicates how often Email/VPIM messages are retrieved from the networked node. See "E-Mail Retrieval Interval (Minutes)" on **page 866** for more information.

VPIM MESSAGES

When VPIM networking is enabled, voice mail subscribers can send outbound voice messages and receive inbound voice messages for subscribers on another voice mail system that is configured for VPIM. VPIM messages are sent and received as follows:

- **Outbound messages**: To leave a voice mail message, the voice mail subscriber calls the voice mail extension and is then prompted to enter the voice mailbox number. When the subscriber enters the mailbox number, the voice processor recognizes the extension as an off-node mailbox. After the subscriber records the message, the voice processor creates an e-mail and attaches the voice message as an audio file attachment. The voice processor routes the e-mail message, via SMTP, to the correct address using the domain information for the remote voice processing platform.

  There are different ways the system sends outgoing VPIM messages:

  - The user on the local voice mail can send messages to the VPIM mailboxes on the remote node using the normal send options within the user's local mailbox. To send an outgoing message from a local mailbox, the mailbox user enters their mailbox and then uses the process of recording and sending messages to a mailbox using the VPIM extension number.

  - On a UVM system, a nonsubscriber internal or external caller that is answered by voice mail, can dial a VPIM mailbox extension and leave messages.

    To leave a voice mail message, the voice mail subscriber calls the voice mail extension and is then prompted to enter the voice mailbox number. When the subscriber enters the mailbox number, the voice processor recognizes the extension as an off-node mailbox. After the subscriber records the message, the voice processor creates an e-mail and attaches the voice message as an audio file attachment. The voice processor routes the e-mail message, via SMTP, to the correct address using the domain information for the remote voice processing platform.

- **Inbound messages**: When the voice processor receives an e-mail message with a voice mail attachment, the mailbox number in the e-mail message specifies to which local mailbox the message belongs. The audio file is removed from the e-mail message and encoded to match the audio format used by the voice processor. The message is then routed to the appropriate subscriber's mailbox and the subscriber is notified that he or she has a new message. From the subscriber's perspective, the message sounds as if it were left locally. The only differences between a message received locally and a VPIM message are:
  - Subscribers with display phones see "UNKNOWN SENDER" rather than the name and extension of the caller when they listen to a VPIM message.
  - Subscribers are not presented with a reply option for VPIM messages.

OVERVIEW OF VPIM PROGRAMMING

To support VPIM, configure the following:

1. **IP Settings** below
2. **Domain Name** below
3. SMTP Server Settings on **page 853**

4. Voice Processing Nodes on **page 854**

5. Off-Node VPIM Mailboxes on **page 854**

6. Mailbox Personalization on **page 856**

For VPIM programming troubleshooting information, see **page 857**.

IP SETTINGS

To configure IP settings for VPIM:

1. Select System – **IP Settings**.

2. Program the following options (refer to the *MiVoice Office 250 DB Programming Help* for details):

   • Base Server Hostname—must match the DNS server hostname (for a system not equipped with a PS-1)

   • Static IP Address

DOMAIN NAME

NOTE	On a system not equipped with a PS-1, the system VPIM messages from the remote location must be directed to the FQDN of the Base Server Hostname. To receive and send e-mail messages using VPIM, the Base Server Hostname in System\IP Settings must be identical to the DNS hostname programmed in the Domain Name field. If the hostname does not match the DNS server hostname or an alias is used for the address, the system cannot resolve the name and its destination, and the VPIM server may reject the message.

To configure the domain name for UVM:

1. Select System – **IP Settings**.

2. In the **Domain Name** text box, type the domain name for your network. The Domain name, hostname, and mailbox number are necessary for the VPIM message process. An example VPIM message is 1000@hostname.domain.com. Where 1000 is the mailbox number, hostname is the destination or recipients server name, followed by the domain name with extension (.com). This message format is used for processing both sending and receiving VPIM messages.

SMTP SERVER SETTINGS

To set the SMTP server:

1. Select System – **E-mail Gateway**.

2. Set the E-mail System to **SMTP**.

3. Program the E-mail SMTP Server option (refer to the *MiVoice Office 250 DB Programming Help* for details).

VOICE PROCESSING NODES

To configure VPIM networking, you need to identify the voice processor that your system will be networked with. To do this, you create and configure nodes for each system in DB Programming under the Voice Processor folder. Keep the following considerations in mind when creating VPIM nodes:

- Each VPIM node has its own an internal message queue (similar to a mailbox's message queues) and stores messages destined for other nodes until they are delivered.

- If some of the fields under a VPIM networked node have a red "X," it is because they do not apply.

- The value in the System Number/Domain field is the Fully Qualified Domain Name (FQDN) of the remote VPIM node (for example, node1.mitel.com).

- If you attempt to change the network type for a remote node to "VPIM" without configuring the SMTP server, an error message appears. You must first program the SMTP Server in the E-mail Gateway folder.

- If you attempt to change the network type from "VPIM" to "None" with one or more VPIM nodes programmed, an error message appears. You must first remove the VPIM nodes.

- The node programming is used for processing outgoing VPIM messages and not incoming. If the system receives a VPIM message and then the node is not programmed, the system will process the VPIM message if the matching mailbox exists.

To create VPIM nodes:

1. Select Voice Processor – Devices – **Nodes**.

2. Right-click and select **Create Node**. After the VPIM node is created, DB Programming shuts down.

3. Reopen DB Programming and go to Voice Processor – Devices – **Nodes**. For the new node's Network Type, select **VPIM**.

4. Configure the **System Number/Domain** field for the node. The value in the System Number/Domain field is the Fully Qualified Domain Name (FQDN) of the remote VPIM node (for example, node1.mitel.com).

OFF-NODE VPIM MAILBOXES

VPIM networking is supported between Mitel voice processors such as UVM. It can also be used to network a Mitel voice processor to a MiCollab Unified Messaging or a third-party voice processor.

To provide VPIM networking functionality for a voice processor, configure the VPIM fields in DB Programming. Then you must create one of the following mailbox types for each subscriber you want to configure for VPIM networking:

- Associated off-node mailboxes: Associated mailboxes can be created when there is a matching extension number on the local or private networking nodes.

This is true for UVM, MiCollab Unified Messaging, and third-party voice mails. If an extension does not already exist on the MiVoice Office 250 network, then the mailbox can be unassociated.

If an existing local or network extension has a mailbox already assigned when creating an associated off-node mailbox, DB Programming indicates this and does not continue.

If an existing local or network extension conflicts with the associated off-node mailbox you are creating, DB Programming provides a dialog box to allow you to associate the mailbox.

The associated off-node mailboxes you create will mirror the existing mailboxes on the local node. When creating VPIM-associated mailboxes under Local, if the associated extension for the mailbox is on Node 3 and not Local, the mailbox is created under the Node 3 folder and not the Local folder. The mailbox is created on the coinciding node for the extension.

- Non-associated off-node mailboxes: Non-associated off-node mailboxes are mailboxes that are not associated to any extension on the MiVoice Office 250 private network.

    Non-associated off-node mailboxes can be used on MiVoice Office 250 systems that are not using private networking, where each system cannot see each others numbering plan.

    This type of mailbox can be used with any voice mail system using VPIM.

    The non-associated off-node mailboxes have the following considerations:

    - *Networked systems*: If the system is using private networking with other MiVoice Office 250 systems, you can program associated and non-associated off-node mailboxes on the local PPXXX node. The nodes listed under the local PPXXX for VPIM can only have associated mailboxes.

    - *Non-networked systems*: If the system is not using private networking with the MiVoice Office 250 systems, then the system allows associated and non-associated off- node mailboxes under the PPXXX VPIM nodes.

VPIM mailboxes act as proxy mailboxes to receive messages from other voice messaging systems.

NOTE	To receive and send e-mail across the Internet to a remote site, UVM needs access outside its local network. The VPIM e-mails are processed on port 25 using SMTP on the MiVoice Office 250. This may require changes to any firewall between the MiVoice Office 250 and the Internet.

To create VPIM associated mailboxes (the originating system is private networked with the destination voice processor):

NOTES	The following instructions assume that you have already imported the device extensions of the system node.
	When creating an associated mailbox, the Remote Mailbox Extension field for the associated mailbox is automatically populated with the extension of the destination mailbox.

1. Select Voice Processor – Devices – Mailboxes – *<VPIM node>* – *<**97XXX (associated mailbox)**>*.

2. Right-click, and then select **Create Associated Off-Node Mailboxes**. A window appears prompting you to select the type of device to include.

3. Select the type of device, and then click **Next**. The list of devices appears. To view items in a list, click **List**.

4. Select the devices for which you want to create mailboxes. Then click **Add Items**. Your selections appear in the mailbox list. Click **Finish** to exit.

5. To select a series of devices, hold down the **SHIFT** key while selecting the first and last devices you want. To select two or more that are not next to each other, hold down the **CTRL** key while selecting the desired devices.

   Extensions on a MiVoice Office 250 that are networked using the private networking feature, but are using a MiCollab Unified Messaging or third party voice mail, are considered associated mailboxes if the off-node mailbox number matches the remote node extension.

To create VPIM non-associated mailboxes (the originating system is not private networked with the destination system or the systems are using private networking, but the extension number does not exist in the numbering plan):

1. Select Voice Processor – Devices – Mailboxes – <*VPIM node*> – **Local**.

2. Right-click, and then select **Create Non-Associated Mailboxes**. A window appears prompting you to select the type of device to include.

3. Enter the extension number you want to use for the first item in the Starting Extension field. Then indicate the number of items you want to create in the Number of Extensions field. The new mailboxes will be assigned sequentially to the next available numbers. When created, they appear in the programming window.

4. Program the Remote Mailbox Extension field to match the extension of the destination mailbox on the local node.

NOTE	When creating a non-associated mailbox, you must manually set the Remote Mailbox Extension field for the non-associated mailbox to match the extension number of the destination mailbox. The Remote Mailbox Extension field is located under mailbox settings in DB Programming. See "Remote Mailbox Extension" on **page 966** for details.

MAILBOX PERSONALIZATION

After you configure VPIM mailboxes in DB Programming, instruct subscribers to personalize their VPIM mailbox by recording a personal greeting and recording their name for the directory. For security reasons, subscribers should change the default mailbox passcode. Refer to the applicable phone user guide for instructions.

For associated off-node mailboxes, the voice mail extension number is the voice mail extension for the *remote* node. For non-associated mailboxes, the voice mail extension number is the voice mail extension for the *local* node.

The associated and non-associated off-node mailboxes used for VPIM have the same feature set as an extension ID mailbox, where a user can record their name, greeting, and change their passcode. VPIM mailboxes do not have the standard mailbox features, such as playing back messages.

TROUBLESHOOTING VPIM NETWORKING

Table 154 includes additional troubleshooting information for VPIM networking issues. In general, you can review the following log files to help troubleshoot VPIM issues:

- **Avdap.log**: Shows the voice mail processes as they pertain to conversion and sending of messages.

- **Mail.mainlog**: Contains messages from the component used to route VPIM messages (EXIM), and includes the source and destination mailbox extensions and any relay errors.

Also see page 858 for additional help with troubleshooting.

Table 154:   VPIM Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
A message did not arrive at the remote node.	The domain name is not correct.	Modify the Domain field in DB Programming to match the FQDN for the VPIM node (see "Domain Name" on **page 853**).
	The destination e-mail username is not correct.	Contact the administrator of the remote node and verify that the E-mail Username field is programmed correctly.
	SMTP server isn't configured for open relaying.	Contact the administrator at the remote node and advise him or her to configure the SMTP server for open relaying.
	The MiVoice Office 250 server's hostname is programmed incorrectly or an alias was used. When a VPIM message arrives, the system cannot resolve the hostname and deliver the message.	To receive and send e-mail messages using VPIM, you must program the hostname for the MiVoice Office 250 Base server under System\IP Settings identically to the hostname you program under Voice Processor\Devices\Nodes*<node>***System Number/Domain**. Verify the FQDN and ensure an alias is not used by looking up the entries in DNS.
	The server name or domain name setting is programmed with the wrong FQDN.	
A message did not arrive at the local node.	A firewall may be blocking port 25. Port 25 must be open for VPIM networking.	Verify that you can ping the system on the appropriate port from an address outside the firewall and network.
	The domain of the sending voice processing platform is not a safe domain.	Be sure that you have added the sending voice processing platform as a remote VPIM node with the correct FQDN.
	Voice mail messages that are greater than 15 to 60 minutes long, will take 30 minutes or longer to process on a UVM system.	Wait more than 30 minutes before checking to see if the e-mail arrived.
The network settings are configured correctly on both VPIM nodes, but messages are not being exchanged.	The VPIM peer is not fully VPIM-compliant.	Capture the e-mails that are being sent between the VPIM peers and compare them against the RFC 2421 Specification.

VPIM Process BVM & EM

Incoming VPIM Message From BVM

Flow Process of the Incoming VPIM Message

Network

Email Gateway
E-Mail SMTP Server
& Email System
Values

Voice Processor Node
Programming.
System Number/Domain and
Network Type

Associated and Unassociated
Mailbox programming
Voice Processing | Devices | Mailbox

BVM or EM user

IP Network

Exchange Server Relay

ExchangeServer.domain.co
m

The VPIM message is
received and process
using Exim Mail from
the External SMTP
Server.
2xxx.hostname.domain

2xxx.hostname.domain
Or
3xxx.hostname.domain

Has the E-Mail
Retrieval Timer
expired

NO → Messages
wait in
queue
Until Timer
Expires

YES

Is the 2xxx or 3xxx
mailbox found in the
under Voice Processing
Node Mailboxes?

NO → Message is not
processed and is
deleted.

YES

Message is
decoded to
G726 or G711

Message is
delivered to the
mailbox

BVM or
EM press
1 to
Listen to
new
messages

Legend: Blue = 5000 Programming Fields In Database Base Programming

NETWORK SETTINGS

The following settings are located under Voice Processor – Networking.

TIMERS AND LIMITS

You can program the following timers and limits for the voice processor networking feature:

- **Failed Connection Retry Interval**: Select the number of seconds that the system will wait between attempts to connect to another Voice Processor. When a connection fails, the system waits this number of seconds and then tries the connection again. The range is 0–32767 seconds; the default value is 60.

- **Failed Connection Timeout**: Select the number of seconds that the system will wait for the other voice processor to respond before the connection attempt is considered a failure. After this timer expires, the system waits until the Failed Connection Retry Interval timer expires before it attempts to connect again. The range is 0–32767 seconds; the default value is 60.

- **Maximum Messages Per Network Call**: Determines the maximum number of messages the voice processor will transmit to a remote node during one call. When this field is programmed to 0, the number of messages per network call is unlimited. Since messages transmitted via e-mail do not tie up any voice processing resources on the voice processor, this field does not apply to e-mail networking. Also, AMIS calls are automatically limited to nine messages per call by AMIS protocol. If this field is programmed between 1 and 9 (inclusive), AMIS calls will adhere to this field. However, if this field is programmed to 0 or something greater than 9, AMIS networking calls can still only transmit a maximum of nine messages. The range is 0–1000 messages; the default value is 0.

- **Maximum Network Call Attempts**: Determines the maximum number of consecutive, unsuccessful call attempts the voice processor will make to a remote node. When call attempts have reached the Maximum Network Call Attempts for a node, all pending messages for the node are returned to the destination programmed in the Undeliverable Messages Destination Type field. See "Undeliverable Messages Destination Type" on **page 861**. No more calls are attempted for the node until the delivery start and stop date and time are active again. The range is 1–99; the default value is 30.

- **Maximum Network Calls**: *UVM only.* Limits the total number of simultaneous network calls. If the voice processor receives a network call when it already has the maximum number of network calls in progress, it will reject the new call. If the voice processor needs to deliver messages to a remote node when it already has the maximum number of network calls in progress, the message delivery will be delayed until the number of current network calls drops below the maximum. This range is 1 to 32; the default is 2.

- **Maximum Network Inbound Connections**: Select the number of inbound connections that the voice processing system will support at one time. This limits the number of TCP/IP networking and e-mail networking protocols used to network voice processing system. The valid range is 0–32767 connections, and the default value is 1.

- **Maximum Network Outbound Connections**: Select the number of outbound connections that the voice processing system will support at one time. This limits the number of TCP/

IP networking and e-mail networking protocols used to network voice processing systems. The range is 0–32767; the default value is 1.

- **Maximum Network Message Length**: Determines the maximum length of a message the voice processor will transmit to a remote node. It is applied as the message is being recorded. When this field is programmed to 0, the message length is unlimited. The range is 1–600 minutes; the default message length is 15 minutes.

- **Network Call Failure Threshold**: Determines the maximum number of attempted calls to a node before a diagnostic alarm is printed. The alarm continues to be printed until the Maximum Network Call Attempts for the node has been reached, or a successful call has been completed. The value of this field can equal the value of the Maximum Network Call Attempts field, but it cannot exceed it. The range is 1–999; the default value is 15.

- **Network Call Retry Timer**: Determines the amount of time the voice processor waits before retrying a network call when the remote site does not answer or the number is busy. The system ignores this timer if the message threshold has been reached. The range is 1–60 minutes; the default is 5 minutes.

- **Update Message Latency Period Timer**: Determines the length of time the AVDAP will wait between update message delivery attempts. When this field is programmed the update message latency time for an update message will be reset. The notification tasks scan each AVDAP node to determine if the message threshold or message latency has expired. If either is true, the AVDAP will spawn a TCP/IP client task which attempts a connection to a remote AVDAP node. However, if there are only update messages in the node queue, the AVDAP checks if the update message latency time has expired when it checks the message threshold or message latency. If the update latency time has not expired, that is, it has not been 15 minutes since the last delivery of updates, a TCP/IP client is not spawned. This range is 1–1440 minutes; the default is 15 minutes

To program a timer or limit:

1. Select Voice Processing – Networking – Timers and Limits – <***timer or limit***>.

2. In the **Value** column, select the value, and then type the new setting in the box.

3. Click out of the field or press **ENTER** to save the change.

VALIDATE OFF-NODE MAILBOXES

The Validate Off-Node Mailboxes option indicates whether or not mailbox numbers are checked against DB Programming when a user attempts to deliver a message through a node network mailbox.

When this option is enabled, the mailbox number entered must be a valid network mailbox number on the local node. This means that the local node can only send messages to mailboxes on the remote node that have network mailboxes programmed on the local node. When this option is disabled, the mailbox number is not checked. This means the local node can send messages to any mailbox on the remote node, even if it does not have a local direct network mailbox.

To enable the Validate Off-Node Mailboxes flag:

1. Select Voice Processing – Networking – Timers and Limits – **Validate Off-Node Mailboxes**.

2. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

3. Click out of the field or press **ENTER** to save the change.

UNDELIVERABLE MESSAGES DESTINATION TYPE

This field is used to determine the destination for all network messages which are undeliverable. It can have a value of Delete, Sender, or System Administrator.

- When the field is set to **Delete** (default), all undeliverable messages are deleted.

- When the field is set to **Sender**, all undeliverable messages are returned to the sender if possible. If the sender is unknown, the messages are returned to the System Administrator's mailbox. If the System Administrator's mailbox does not exist, the messages are deleted.

- When the field is set to System Administrator, all undeliverable messages are returned

To set the Undeliverable Messages Destination Type:

1. Select Voice Processing – Networking – Timers and Limits – **Undeliverable Messages Destination Type**.

2. In the **Value** column, select the option from the list.

3. Click out of the field or press **ENTER** to save the change.

VOICE PROCESSOR SYSTEM SETTINGS

Voice processor system settings apply to UVM systems only. If you are using a MiCollab Unified Messaging system, see **page 836** for system settings information.

Voice Processor system settings include the following:

- Dial-0 Destinations on **page 863**

- Total Storage Disk Usage Statistics on **page 863**

- System Administrator Mailbox on **page 863**

- Alternate Tone Detection on **page 864**

- Volume on **page 864**

- Save Message on Return Call on **page 864**

- Swap "7 for Save" and "9 for Delete" Message Keys on **page 865**

- Identification Prompt on **page 865**

- Monitor Password on **page 866**

- BS-UVM System Recording Codec on **page 866**

You can view or program voice processor system settings from the Voice Processor level in DB Programming, as shown in the figure below.

NOTE	"Windows Networking" settings apply to VPU systems only and are no longer supported.
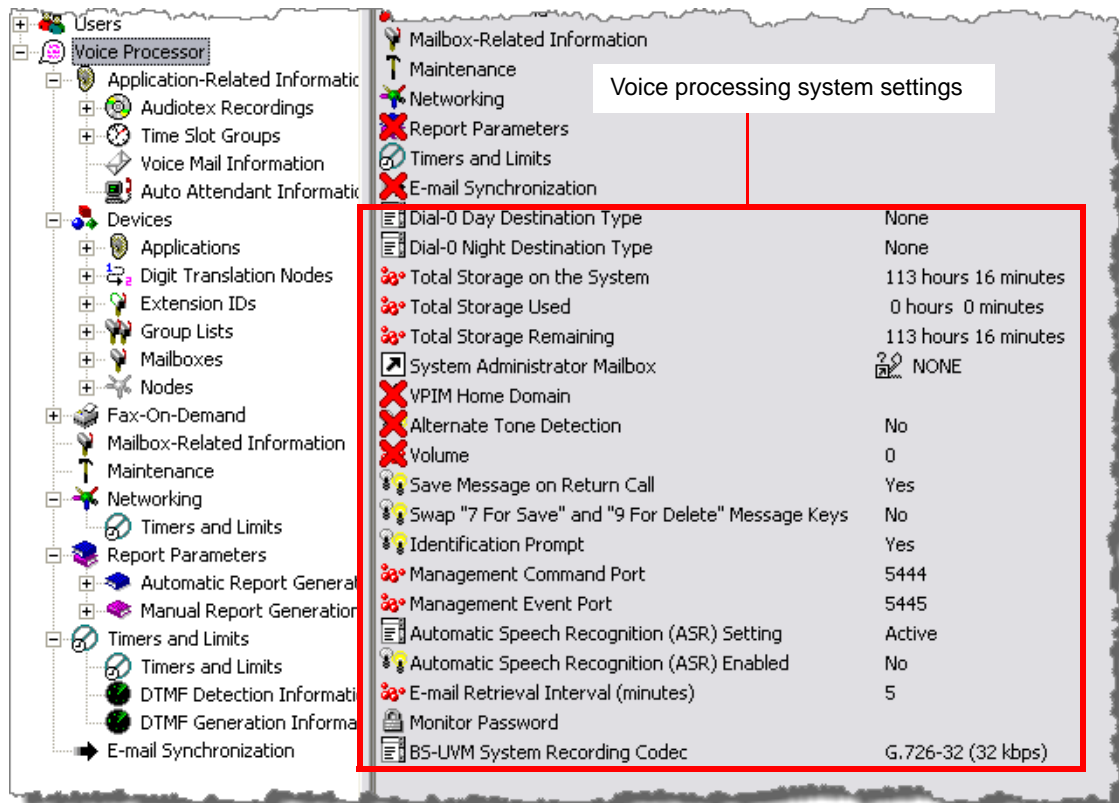
DIAL-0 DESTINATIONS

System Dial-0 Destination options are the same as mailbox Dial-0 Destination options. See **page 978**.

TOTAL STORAGE DISK USAGE STATISTICS

Total Storage options show voice processing computer disk storage status. The information is presented for reference only and cannot be programmed. Disk Usage Statistics include the following:

- **Total Storage on the System**: The amount of disk space (in hours) available on the voice processing computer.

- **Total Storage Used**: The amount of disk space (in hours) that has been used by the voice processing applications.

- **Total Storage Remaining**: The amount of disk space (in hours) remaining in the voice processing computer. If the disk is nearly full, you can increase disk space by clearing the mailbox messages. If available disk space is frequently low, you should install a larger-capacity disk drive in the voice processing computer.

To view Disk Usage Statistics:

Select **Voice Processor**. Disk Usage Statistics are shown in the right pane.

SYSTEM ADMINISTRATOR MAILBOX

The system administrator mailbox is used to:

- Record custom audiotex recordings for voice processing applications.

- Send broadcast messages to all subscribers.

- Perform mailbox and group list maintenance.

For more information about using the administrator mailbox, refer to the *MiVoice Office 250 Unified Voice Messaging Administrator Guide,* part number 580.8009.

To assign the System Administrator Mailbox:

1. Select Voice Processor – **System Administrator Mailbox**.

2. In the **Value** column, right-click the existing value, and then select **Change System Administrator Mailbox**. A window appears prompting for the device type to include.

3. Select **Mailbox**, and then click **Next**. The items with details appear. To view items in a list only, click **List**.

4. Select the desired mailbox, and then click **Finish**. The selection appears in the System Administrator Mailbox field.

ALTERNATE TONE DETECTION

In some remote messaging situations, the device being called uses signaling tones that the voice processor does not recognize, at default. If remote messaging is not properly signaling the device, enable this option to allow the voice processor to detect alternate tones.

To enable the Alternate Tone Detection option:

1. Select Voice Processor – **Alternate Tone Detection**.

2. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

3. Click out of the field or press **ENTER** to save the change.

VOLUME

> **NOTE** The Volume field is not supported by UVM.

You can set at the same time the volume level for all of the voice channels used by the voice processor. This includes all prompts, voice mail messages, and audiotex recordings. These options cannot be adjusted separately and only the playback volume is affected, not the recording volume.

When a voice mail user increases or decreases the volume during the call, the system volume level currently programmed does not change. Only the voice channel being used by the caller is temporarily altered. When the user has completed the call, the system resets the volume of the voice channel used to the selected setting. The volume range is -8 for the softest setting to +8 for the loudest. The normal setting is 0.

To set the volume level:

1. Select Voice Processor – **Volume**.

2. In the **Value** column, enter the new value in the box.

3. Click out of the field or press **ENTER** to save the change.

SAVE MESSAGE ON RETURN CALL

The Return Call feature allows a voice mail user to place return calls to message senders (with Caller ID or extension numbers) directly from the voice mailbox. The Save Message on Returned Call flag tells the voice processor whether or not to save the message that the user is listening to at the time the user chooses to make the return call.

If the flag is set to **Yes**, the voice processor automatically moves the message to the user's saved message queue when the user places the return call. If the message is already in the saved message queue when the user places the return call, then it is unchanged. This applies to both the new message queue and deleted messages. If the flag is enabled and the user places a return call from a deleted message, that message is moved to the saved message queue, restoring the message. If the flag is set to **No**, the voice processor leaves the message in the same queue it was in before the user placed the return call. Note that if the flag is set to

No and the message is in the new message queue, the voice processor refreshes the message lamp on the user phone.

To enable the Save Message On Return Call option:

1. Select Voice Processor – **Save Message on Return Call**.

2. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

3. Click out of the field or press **ENTER** to save the change.

SWAP "7 FOR SAVE" AND "9 FOR DELETE" MESSAGE KEYS

You can swap the 7 and 9 keys for all mailboxes. This changes the default value for any *new* mailboxes. All existing mailboxes are not changed. You must manually change existing mailboxes. For more about the feature and individual mailbox programming instructions, see **page 989**.

To swap the 7 and 9 message keys:

1. Select Voice Processor – **Swap "7 for Save" and "9 for Delete" Message Keys**.

2. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

3. Click out of the field or press **ENTER** to save the change.

IDENTIFICATION PROMPT

The Identification Prompt option enables or disables a custom message that identifies the voice processing system. When enabled, this prompt is heard before the voice mail greeting and states, "Your call is being handled by the voice processing system." This prompt automatically plays when external callers:

- Are transferred to a voice mailbox. The system plays the ID prompt before playing the mailbox greeting (primary, alternate, or system).

- Are forwarded to a voice mailbox. The system plays the ID prompt before playing the mailbox greeting (primary, alternate, or system).

If the identification prompt is disabled, it is not played before the voice mailbox greeting when an external caller accesses a mailbox. By default, this flag is enabled.

To disable the Identification Prompt:

1. Select Voice Processor – **Save Message on Return Call**.

2. In the **Value** column, clear the check box. The field changes to **No**. To enable the feature, select the check box.

3. Click out of the field or press **ENTER** to save the change.

E-MAIL RETRIEVAL INTERVAL (MINUTES)

For UVM systems only. This option indicates how often Email/VPIM messages are retrieved from the networked node. The range is 1–60 minutes; the default is 5 minutes. For example, the UVM receives 4 VPIM messages for the same mailbox at or about the same time. The UVM processes one message every 5 minutes, the first message in 5 minutes, the second message in 10 minutes, the third message in 15 minutes and the fourth message in 20 minutes. This option appears with a red "X" for voice-processing systems other than UVM.

To set the E-Mail Retrieval Interval (minutes) option:

1. Select Voice Processor – **E-Mail Retrieval Interval (minutes)**.

2. In the **Value** column, select the option from the list.

3. Click out of the field or press **ENTER** to save your changes.

MONITOR PASSWORD

To protect the voice processor against unauthorized access, you can require a password for the AVDAPMon utility.

To program the password:

1. Select Voice Processor – **Monitor Password**.

2. Right-click **Monitor Password**, and then select **Edit Password**. The Edit Monitor dialog box appears.

3. Enter the current password, if one exists.

4. Type the new password (up to 40 characters) in the New password box. Typed characters appear as asterisks (***).

5. Retype the password in the **Confirm password** box.

6. Click **OK** to exit and save the password. If the entered passwords match, you return to the Password field. If not, you must re-enter the new password and verify it again.

NOTE	To provide system security, the e-mail system must have a password. To make the passwords difficult to guess, they should not consist of predictable patterns, such as one digit repeated several times.

BS-UVM SYSTEM RECORDING CODEC

This audio enhancement for system recordings applies to Base Server (BS) UVM systems only.

For BS-UVM systems, you can use either the default G.729A codec or the G.726-32 codec.

G.726-32 codec recordings offer higher voice quality and result in a higher Mean Opinion Score (MOS), especially if other G.729 codecs are in the call path. However, G.726-32kbps recordings require four times the space on the compact flash-type memory card to store the same recording lengths. If you use the G.726-32 codec, you may want to upgrade to a larger compact flash-type memory card.

NOTE	Selecting the G.726-32 codec prompts UVM reset.

To program the system recording codec:

1. Select Voice Processor – **BS-UVM System Recording Codec**.

2. In the **Value** column, select the option from the list.

3. Click out of the field or press **ENTER** to save your change. If you selected G.726-32, a message appears to notify you that changing the codec to G.726-32 does the following:

   • Causes UVM to reset.

   • Affects the storage space on the compact flash-type memory card.

   • Does not affect all previous recordings that use the G.729A codec.

4. Click **Yes** to proceed. UVM is reset.

TIME SLOT GROUPS

Applies to UVM voice processors only. Each application is assigned to a "Time Slot Group," which determines the maximum number of voice channels (up to 32) used by the applications in that group. Voice channels are used for processing calls between the system and the applications.

The programmed voice channel limit for the time slot group may exceed the number of voice channels actually provided by the hardware, due to hardware limitations and/or heavy system traffic. For example, if an Automated Attendant application is assigned to a time slot group that has a programmed limit of five voice channels, it can normally support five simultaneous transfers of outside calls to extensions. However, if only four voice channels are available (because all other channels are in use by other applications or the hardware only supports four channels), a fifth call cannot be completed to that (or any other) application. When all voice channels are busy, intercom callers hear reorder tones and see a CALL CANNOT BE COMPLETED display; outside callers hear ringing, but their calls are not answered.

Because multiple voice channels can be made available to the applications, callers are unlikely to encounter busy resources. In addition, the use of applications as recall destinations and attendants make it even less likely that a caller will not reach an available resource. For example, an Automated Attendant application can have one Automated Attendant Recall Destination with several voice channels as its recall destination and another Automated Attendant Recall Destination application (with several voice channels) as its attendant. Recalls from that Automated Attendant application would be very likely to reach an available voice channel at one of the Recall Destination applications.

AUTOMATED ATTENDANT

Recalls	Attendant
First Automated Attendant Recall Destination application	Second Automated Attendant Recall Destination application
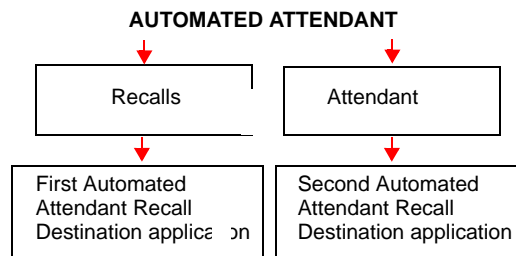
Call Routing Announcement digit translation nodes do not have assigned time slot groups.

CHANGING TIME SLOT DESCRIPTIONS

The Time Slot Group list shows the time slot groups and the number of channels currently assigned to the groups.

To change the description of a time slot group:

1. Select – Voice Processor – Application-Related Information – **Time Slot Groups**.

2. Click once on the current description.

3. Type the new description in the box, using up to 20 characters.

4. Click out of the option or press **ENTER** to save the change.

CHANGING TIME SLOT MAXIMUM CHANNEL ALLOCATIONS

The **Maximum Channel Allocation** determines the maximum number of voice channels that are used by the applications assigned to the time slot group. By default, this field displays the maximum available voice channels, as determined by the hardware configuration.

The combined total of voice channels assigned to all time slot groups may exceed the number of voice channels provided by the hardware. This is allowed because it is not likely that all time slot groups will be busy at once. However, an individual time slot group is limited to the Maximum Channel Allocation. This parameter is used for setting the limit of voice channels that will be used by the associated application.

Increasing UVM Ports

Any time you increase the number of UVM ports, you may need to adjust the voice processing Time Slot Groups channel allocation in DB Programming. All Time Slot Groups default to four channels; however, the maximum number of channels allowed is equal to the number of licensed UVM ports.

Beyond possibly adjusting the time slot channel allocation, there is no other special programming required to support additional voice mail ports. To verify the licensed port numbers on your system, see the Unified Voice Messaging Ports value under the Software License folder (see "Software License Descriptions" on **page 50**).

To change the **Maximum Channel Allocation** *of a time slot group:*

1. Select – Voice Processor – Application-Related Information – **Time Slot Groups**.

2. In the **Maximum Channel Allocation column,** enter the maximum number of voice channels to be used by the applications.

VOICE PROCESSOR APPLICATIONS

UVM systems only. Voice Processing can be used for any of the following applications:

- **Automated Attendant: The Automated Attendant** Provides automated call answering services. Calls can be transferred, forwarded, or directly ring-in to an automated attendant. When an automated attendant answers a call, the application plays a recording that gives the caller dialing instructions to reach their destination. After hearing the recording, or at any time while it is playing, the caller may dial an extension or mailbox number. See **page 874**.

- **Automated Attendant Recall Destination:** If a call transferred by an automated attendant is not answered before the system's Transfer Voice Processor timer expires, the call recalls the Automated Attendant Recall Destination. The Recall Destination announces that the called phone is unavailable and allows the caller to leave a message, if the phone has an associated mailbox, or dial another extension. See **page 878**.

- **CRA:** CRAs can be used two ways:

  - A CRA application can be used in place of a playback device. The playback device function is especially useful for programming hunt group announcement and overflow phones. When called, the CRA application plays a recording and then hangs up.

  - The CRA application can use Digit Translation which allows the caller to press a single digit for access to a mailbox, a Fax-On-Demand function, or a phone or hunt group that has an associated mailbox or extension ID. Digit translation can be programmed for each digit 0–9, #, and *, plus a Timeout that is used when the caller does not enter a digit. Each digit can lead to a "digit translation node" that has its own digit translation values. This layered CRA digit translation creates a "tree" of programmable digit translation nodes. See **page 879**.

- **Message Notification/Retrieval:** Only one Message Notification/Retrieval application may be created and it *must* be created to allow Voice Mail message notification and quick mailbox access. See **page 890**.

- **Record-A-Call:** This feature allows a phone user to record an ongoing call in a mailbox message. When a phone user enters the Record-A-Call feature code, the system places a call to the assigned Record-A-Call application for that phone. When the application answers, the system sets up a conference call with the Record-A-Call mailbox for that phone. At default, the mailbox plays a greeting to indicate that the recording is in progress. See **page 893**.

- **STAR:** The Scheduled Time-based Application Router (STAR) enhances the programmability of the Voice Mail application greetings. With STAR, applications can be programmed to play alternative greetings for holidays and weekends. A STAR application is a table of up to 20 entries that serves as a "routing table," which tells Voice Processing which application is used, based on day and time information in the table. The applications are programmed to play the greetings, not the STAR application. The STAR routes the call to the application identified in DB Programming. See **page 911**.

- **Voice Mail:** This application handles all calls that are directed to Voice Mail (other than through the Message Notification/Retrieval application). Callers hear the main company greeting, followed by a menu of available options. Phones can forward or transfer calls directly to their mailbox using this application. See **page 915**.

- **Directory Services:** Directory services provide callers with a list of mailboxes and extension IDs. See "Directories" on **page 926**.

The following are voice processing applications:

- Voice Processing Application Options on **page 872**

- Auto Attendant Recall on **page 878**

- Call Routing Announcements on **page 879**

- Message Notification/Retrieval on **page 890**

- Record-A-Call on **page 893**

- Voice Mail (Application) on **page 915**

CREATING VOICE PROCESSOR APPLICATIONS

You can create up to 150 voice processor applications, but there can be only one Message Notification/Retrieval application. In the default database, extension numbers 2500–2649 are reserved for applications, but any available extension number can be used.

> **NOTE** Complete Time slot group and voice channel programming before creating and programming applications. See **"Time Slot Groups"** on **page 868**.

To create a new application for a Voice Mail using UVM:

1. Select Voice Processor – Devices – **Local** ("Local" is applicable only if a remote node is connected).

2. Double-click **Applications**.

3. Right-click in a blank area of the right pane. A list of options appears to create applications. After you create an application, you cannot change the application type. You can, however, delete it by selecting it, right-clicking and selecting **Delete**.

4. Type the description (up to 20 characters) and user name (up to 16 characters) for each application. The description appears wherever the application is listed and in the system directory. The username appears on phone displays when the application is used.

5. Click out of the field or press **ENTER** to save the change.

To create a new application for a SIP Voice Mail using MiCollab Unified Messaging:

See "SIP Voice Mails" on **page 711**.

CHANGING MULTIPLE APPLICATION EXTENSIONS

You can change several voice processing application extensions at the same time.

To change several extensions at the same time:

1. Select Voice Processor – Devices – **Local** (the "Local" option is shown only if a remote node is connected).

2. Double-click **Applications**.

3. Select the applications that you want to change. You can use the CTRL and SHIFT keys to select multiple applications.

4. Right-click, and then select **Batch Extension Change**.

5. In the Create Extension dialog box, select the starting extension number. The other selected applications are numbered consecutively after this number.

6. Click **OK** and the extensions are automatically renumbered and resorted in the list.

COPYING AND PASTING APPLICATION ATTRIBUTES

The copy and paste operations in DB Programming allow you to copy certain attributes from one application and paste the attributes into several other like applications. This allows you to quickly program several similar devices at one time.

To copy an application:

1. Select Voice Processor – Devices – **Local** (the "Local" option is shown only if a remote node is connected).

2. Double-click **Applications**.

3. Right-click the application extension, and then select **Copy**.

4. To paste the programming information into another application, right-click the application where you want to paste the information, and then select **Paste**. The Copy dialog box allows you to select the attributes you want to copy.

5. Select the desired attributes, and then click **OK**.

VOICE PROCESSING APPLICATION OPTIONS

After creating voice processing applications, you must program the options for each application. **Table 155** shows options you can program for voice processing applications.

Table 155: Voice Mail Application Programming Fields

FIELD	REF. PAGE	AA[1]	AAR	CRA	MNR	RAC	STAR	VM
Digit Translation	**884**			✓				
Greetings	**915**	✓		✓		✓		✓
Accept AMIS Calls	**890**			✓[2]				
Class of Service	**890**				✓			
Attendant	**916**	✓	✓	✓	✓			✓
Auto Fax Detection – Fax Delivery Destination – Fax Delivery E-Mail Address	**888**			✓[3]				

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Table 155: Voice Mail Application Programming Fields (continued)

FIELD	REF. PAGE	AA[1]	AAR	CRA	MNR	RAC	STAR	VM
Music-On-Hold Parameters – Audio for Calls Camped onto this Device – Audio for Calls Holding for this Device – Audio for Calls Ringing this Device	**918**	✓	✓	✓	✓	✓	✓	✓
Time Slot Group	**917**	✓	✓	✓	✓	✓	✓	✓
Transfer Recall Destination	**918**	✓	✓	✓	✓			✓
Schedules							✓	
Default Application 1-20	**914**						✓	
Propagate Original Caller ID on Transfer	**919**				✓			
Calling Party Name and Number	**919**				✓			

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1. Your Automated Attendant application may also require Extension ID programming for phones that do not have mailboxes.
2. If the system is running Unified Voice Messaging, these fields are marked with a red "X."
3. If the system is running Unified Voice Messaging, these fields are marked with a red "X."

AUTO ATTENDANT

This section contains the following information:

- **Feature Description** below

- Viewing Auto Attendant Information on **page 875**

- Programming Auto Attendant on **page 876**

FEATURE DESCRIPTION

UVM systems only. Automated Attendant is a programmable application that provides automated call answering services. Calls can be transferred, forwarded, or directly ring-in to an automated attendant. Calls to the automated attendant application are processed as shown in the following flow diagram.

CALL TO AUTOMATED ATTENDANT
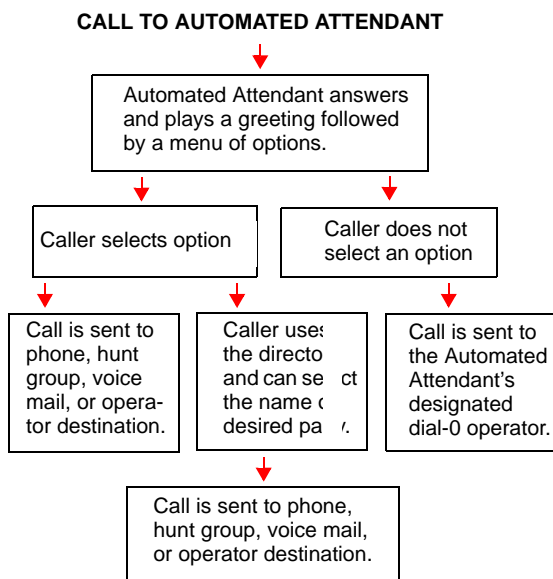
When an automated attendant answers a call, it plays a recording that gives dialing instructions. During or after the recording, the caller may then directly dial a phone extension number, Voice Mail access number (if there is no associated mailbox), or hunt group pilot number. Or, the caller may use the directory to look up the desired extension.

In a network setting, a trunk on another node can ring in to a Voice Processing application.

When the automated attendant answers a call, the caller hears the company greeting, followed by instructions and the list of available options. The caller then has the following options:

- **Dial a phone extension number**: If an extension number is dialed, the call is transferred to the selected phone. If ringback tones are enabled, the caller hears ringing while the call is being transferred. If ringback is not enabled, the caller hears music. If the called phone is forwarded, the call follows the programmed forward.

- **Dial a hunt group number**: When a hunt group number is dialed, the call is transferred to the selected hunt group. The call rings or circulates according to how the hunt group is

programmed (linear or distributed). If ringback tones are enabled, the caller hears ringing while the call is being transferred.

- **Dial the Voice Mail application's extension number**: The caller can reach the Voice Mail main greeting by dialing the application extension number (access number) assigned to the Voice Mail feature. The caller can then leave a message as a non-subscriber or access any of the Voice Mail subscriber features. For more information, see **page 915**.

- **Use the directory**: If the caller does not know the extension or mailbox number of the desired party, he or she can spell the name using the keypad buttons and "look up" the number in the directory. (This option can be disabled in the database. Or, if there are no names recorded for the individual mailboxes or for the system's extension IDs, this option is not provided.) Directory names can be sorted by first or last name. (See **page 926** for information about using the directory.)

- **Dial the operator access destination**: If the caller needs further assistance, dialing 0 accesses the voice processor programmed operator destination. Or, if the caller is on a rotary telephone and cannot enter a digit, the call is automatically transferred to the operator destination. (The operator access destination is programmed in the database. There can be separate destinations for day and night modes.)

NOTE	Due to the natural characteristics of the trunk, the volume level of DTMF tones transmitted over the trunk may be substantially reduced before reaching the phone system and the voice processor. This natural degradation in tone volume may adversely affect the reliability of the Automated Attendant feature. Other factors which can affect automated attendant performance are trunk noise and the quality and strength of the DTMF tones generated by the off-premises phone itself.

AUTOMATED ATTENDANT USES

Two common uses for the automated attendant application include:

- Direct ring-in calls to a busy attendant could be forwarded to an automated attendant by programming the Call Forward If Busy feature or the system forwarding feature.

- Calls could ring in directly to the extension number of an automated attendant application when the system is in day mode or night mode.

AUTOMATED ATTENDANT CUSTOM AUDIOTEX RECORDINGS

Custom audiotex recordings are made using the Voice Mail System Administrator's mailbox. Each recording is associated with a recording number and assigned to the application(s) in database programming or using the System Administrator's mailbox. For more information about creating custom audiotex recordings, refer to the *MiVoice Office 250 Voice Mail Administrator Guide,*.

For information about programming voice processing application options, see **page 915**.

VIEWING AUTO ATTENDANT INFORMATION

You can enable or disable the automated attendant directory and determine whether the directory is sorted by first or last name using these fields. You can also determine whether the system automated attendant transfer prompt plays when calls are transferred to phones without mailboxes or extension IDs.

PROGRAMMING AUTO ATTENDANT

This section contains the following programming information:

- Auto Attendant Directory on **page 876**
- Auto Attendant Transfer Prompt on **page 876**
- Auto Attendant Directory Sort Order on **page 876**
- Auto Attendant Transfer Method on **page 877**

AUTO ATTENDANT DIRECTORY

If the Auto Attendant Directory is enabled, the system allows anyone routed to an Auto Attendant application to access the company directory. If the option is disabled, the Auto Attendant does not prompt the caller for the directory option while in the Auto Attendant area.

To enable or disable the Auto Attendant Directory:

1. Select – Voice Processor – Application-Related Information – Auto Attendant Information – **Enable Auto Attendant Directory**.

2. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

3. Click out of the option or press **ENTER** to save the change.

AUTO ATTENDANT TRANSFER PROMPT

The Auto Attendant Transfer Prompt determines if the transfer prompt ("Please hold while your call is being transferred to...") plays after a caller has entered an extension number that does not have an associated mailbox or extension ID. This applies to transfers from CRA applications that use the "Transfer To Extension" action, which is described on **page 884**.

To enable or disable the Auto Attendant Transfer prompt:

1. Select – Voice Processor – Application-Related Information – Auto Attendant Information – **Auto Attendant Directory**.

2. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

3. Click out of the option or press **ENTER** to save the change.

AUTO ATTENDANT DIRECTORY SORT ORDER

This option is only for Auto Attendant calls. The Auto Attendant Directory Sort Order determines if the mailbox and extension ID descriptions in the directory will be sorted by first name or last name. To change the Voice Mail Directory Sort Order, see "Changing the Voice Mail Directory Sort Order" on **page 929**.

To change the auto attendant directory sort order:

1. Select – Voice Processor – Application-Related Information – Auto Attendant Information – **Directory Sort Order**.

2. From the **Value** column, select the option from the list.

3. Click out of the option or press **ENTER** to save the change.

AUTO ATTENDANT TRANSFER METHOD

Determines how transfers will be made to other applications from Call Routing Announcement applications. The options are Announce Only, Screened, and Unannounced (the default setting is Unannounced). For more information about transfer methods, see "Call Screening" on **page 279**.

To set the Auto Attendant transfer method:

1. Select – Voice Processor – Application-Related Information – Auto Attendant Information – **Auto Attendant Transfer Method**.

2. From the **Value** column, select the option from the list.

3. Click out of the option or press **ENTER** to save the change.

AUTO ATTENDANT RECALL

When a phone receives a call that has been routed through the automated attendant, the call is handled as a transferred call, and the display shows TFR FROM <description>. If the call is not answered before the Transfer Voice Processor timer expires, the call recalls the automated attendant's recall destination. The recall destination is usually the Automated Attendant Recall Destination application which announces that the called phone is unavailable and allows the caller to choose to leave a message, if the phone has an associated mailbox, or dial another extension number.

CALL TO AUTOMATED ATTENDANT

Automated Attendant answers and transfers the call to the destination selected by the caller.

The party receiving the transfer does not answer before the Transfer Voice Processor timers expires.

Call goes to the Automated Attendant Recall Destination where caller selects option.

Call is sent to the phone associated mailbox

Call is sent to phone, hunt group, voice mail, or operator destination.

If the Recall Destination fails to answer a call, it is automatically sent to the recall destination programmed attendant. If the call is not answered there, it is disconnected after the Abandoned Call timer expires.

If an invalid number is dialed, the caller is prompted to enter another number. If a caller does not make an entry before the Inactivity Alarm timer expires, the caller is prompted again to make an entry.

The caller cannot access trunks or enter feature codes through the Automated Attendant application. Trunk access codes and feature codes are considered invalid numbers.

To program Auto Attendant Recall options:

1. Select Voice Processor – Devices – **Local** ("Local" is applicable only if a remote node is connected).

2. Double-click **Applications**.

3. Double-click **Auto Attendant Recall**.

4. Program the options described "Day and Night Greetings" on **page 915**.

CALL ROUTING ANNOUNCEMENTS

This section contains the following information:

- **Feature Description** below

- Programming a Call Routing Announcement on **page 883**

- Multilingual Capability on **page 888**

FEATURE DESCRIPTION

The Call Routing Announcement (CRA) application can be used as a simple playback device that plays a message and then hangs up to disconnect the call. This capability is especially useful for programming hunt group announcement and overflow stations. Or, the CRA application can use digit translation to allow the caller to press a single digit for access to a phone, hunt group, or voice mailbox.

**CALL TO CRA
WITHOUT DIGIT TRANSLATION**

↓

CRA answers and plays a recording.

↓

CRA hangs up. (If this is an announcement or overflow phone, the call returns to the hunt group.)

**CALL TO CRA
WITH DIGIT TRANSLATION**

↓

CRA answers and plays a greeting followed by a menu of options.

↓ ↓

Caller selects option

Caller does not select an option (Timeout)

↓ ↓

Call is sent to phone, hunt group, voice mail, operator destination, digit translation node, or directory as determined by digit programming.

Call is sent to phone, hunt group, voice mail, operator destination, digit translation node, or directory as determined by Timeout programming.

The system supports a Primary and a Secondary Language.For more information, see "Multilingual Capability" on **page 888**.

NOTE	52xx/53xx-series IP phones do support the Japanese language. For complete information about multilingual capability, see "Multilingual Capability" on **page 314**.

When a CRA application is used as a hunt group announcement or overflow phone, calls to the application automatically stop circulating through the hunt group if the caller selects a valid digit translation option. This allows the application to send the call to other phones without the call being "pulled back" into the hunt group when a hunt group phone becomes available and answers the call. However, if the caller does not dial a valid digit translation option, the call is pulled back if a hunt group member answers. An example of a CRA tree is shown on the next page.

Call Routing Announcements (CRAs) can be used two ways:

- A CRA can be used in place of a playback device. The playback device function is especially useful for programming hunt group announcement and overflow phones. When called, the CRA application plays a recording and then hang up.

- The CRA can use Digit Translation (see **page 884**), which allows the caller to press a single digit for access to a mailbox, a Fax-On-Demand function (external voice processing systems only), or to a phone or a hunt group that has an associated mailbox or extension ID. Digit translation can be programmed for each digit 0–9, #, and *, plus a Timeout that is used when the caller does not enter a digit. Each digit can lead to a "digit translation node" that has its own digit translation values. This layered CRA digit translation creates a "tree" of programmable digit translation nodes. There can be up to 200 digit translation nodes in the system and up to 20 for each CRA application.

Use the following guidelines to design an effective CRA application:

- Design with the caller in mind, not just the information you want to include.

- Keep menus as simple as possible, with four or fewer options per menu.

- Number options sequentially and do not skip numbers. List "transfer to operator" last.

- Use consistent digits for options, such as 1 for Yes, 2 for No, and 0 for the operator.

- State the option before the digit. For example, say, "For account information, press 1," instead of, "Press 1 for account information."

- Draw a map of your arrangement to avoid "dead ends" or endless loops.

- Take advantage of Caller ID [CLID] and DNIS to route calls to suitable menus.

- Keep recordings short (under 60 seconds) and do not use jargon.

- Give the most frequently requested information in the first 10 seconds, without requiring the caller to press a digit.

- Make sure the recordings are clear and the voice is consistent from prompt to prompt. Avoid heavy regional accents.

- Do not repeat the main greeting on any other level.

- Make seasonal changes when necessary, but keep menu options the same because callers get used to them. Tell the caller if option changes have been made.

- Include an option for overriding the Primary Language. For example, say, "For English, press 1. Para Español, empuje 2."

- Test your application any time you make a change. Listen to your prompts periodically.

EXT 2500: "*Thank you for calling X Company, if you know the party's extension number, you may dial it now. For a company directory, press #. For Sales information press 3. For technical support, press 4.*"

CALLER DIALS #	*CALLER DIALS NUMBER BEGINNING WITH 1*	*CALLER DIALS NUMBER BEGINNING WITH 2*	*CALLER DIALS 3*	*CALLER DIALS 4*
Company Directory	Call is sent to extension number dialed	**Transfer to Sales Node:** "*To speak to the operator, press 0. To select Sales information by fax, press 1. To speak to a Sales representative, press 2.*"		Transfer to Hunt Group 2001 (Tech Support)

CALLER DIALS 0	*CALLER DIALS 1*	*CALLER DIALS 2*
Transfer to Operator	**Transfer to Node (Sales Info Fax):** "*If you know the fax document number please dial it now. If you wish to speak to the operator, dial 0. For a product list, dial 2. For a price list, dial 3. For training information, dial 4. Then press # to check your selections and enter your fax number or press * to cancel.*"	**Transfer to Extension** 2000 (Sales Hunt Group)

CALLER DIALS 0	**Transfer to Operator**
CALLER DIALS 1997, 1998, or 1999	**Fax with that number is selected**
CALLER DIALS 2	**Selects Fax Document 1999 - Product List**
CALLER DIALS 3	**Selects Fax Document 1998 - Price List**
CALLER DIALS 4	**Selects Fax Document 1997 - Training**
CALLER DIALS #	**End Fax Selections:** System requests dialing information and sends fax, then returns to Main Menu
*CALLER DIALS ***	**Cancel Fax Selections: System returns to greeting**

Use the following guidelines to design an effective CRA application:

- Design with the caller in mind, not just the information you want to include.

- Keep menus as simple as possible, with four or fewer options per menu.

- Number options sequentially and do not skip numbers. List "transfer to operator" last.

- Use consistent digits for options, such as 1 for Yes, 2 for No, and 0 for the operator.

- State the option before the digit. For example, say, "For account information, press 1," instead of, "Press 1 for account information."

- Draw a map of your arrangement to avoid "dead ends" or endless loops.

- Take advantage of Caller ID [CLID] and DNIS to route calls to suitable menus.

- Keep recordings short (under 60 seconds) and do not use jargon.

- Give the most frequently requested information in the first 10 seconds, without requiring the caller to press a digit.

- Make sure the recordings are clear and the voice is consistent from prompt to prompt. Avoid heavy regional accents.

- Do not repeat the main greeting on any other level.

- Make seasonal changes when necessary, but keep menu options the same because callers get used to them. Tell the caller if option changes have been made.

- Include an option for overriding the Primary Language. For example, say, "For English, press 1. Para Español, empuje 2."

- Test your application any time you make a change. Listen to your prompts periodically.

To program a Call Routing Application:

1. Read the guidelines above before continuing.

2. Create the Call Routing Application and enter its description and user name.

3. Change the day and night greetings to custom recordings. Remove the default recordings and assign new recording numbers. Write down the recording numbers and their assignments for later use.

4. If using Digit Translation, create the digits and nodes you are using.

   - Give them appropriate descriptions and usernames. For example: Sales 1, Service 2, Repair 3, and so forth.

   - Go back to the Applications Programming screen and expand the Call Routing application that you have just created.

   - Program each digit or node individually and set the greetings as desired. Repeat this step for all nodes and digits.

5. Use the System Administrator's mailbox to record your custom greetings. For more information, refer to the refer to the *MiVoice Office 250 Voice Mail Administrator Guide,*.

6. Test the application and all of its nodes by calling the application and checking each digit and node.

CRA CUSTOM AUDIOTEX RECORDINGS

When accessed, the CRA application plays a custom audiotex recording and then hang up.

CRAs are recorded using the Voice Mail System Administrator's mailbox. For more information, refer to the *MiVoice Office 250 Voice Mail Administrator Guide,*.

Each announcement is assigned to use a "custom greeting" number in database programming.

A CRA application message can be programmed to include the caller's queue position and/or estimated wait time. The queue position announcement tells a caller how many calls are ahead of their call. This includes calls being served and waiting calls.

Recordings for fax documents should include all dialing instructions. The recording should state if documents can be selected by number and/or list all options. If fax delivery times are set to specific days or times, the recording should also include this information.

CALL SCREENING

Calls transferred from the Automated Attendant or a CRA application can be screened, announced, or unannounced. Separate programming flags determine the methods used for transferring calls to phones with mailboxes, phones with extension IDs, and extensions without mailboxes or IDs. See **page 920** for details about extension IDs.

PROGRAMMING A CALL ROUTING ANNOUNCEMENT

Make sure you read the guidelines on **page 879** before you program the CRA.

To program a CRA:

1. Create the CRA (see **page 871**).

2. Change the day and night greetings to custom recordings. Remove the default recordings and assign new recording numbers. Write down the recording numbers and their assignments. You will record them later.

3. If you are using Digit Translation, create the digits and nodes you are using as follows.

   a. Appropriately name the descriptions and usernames. For example: Sales 1, Service 2, Repair 3, and so forth.

   b. Return to the Applications Programming screen and expand the Call Routing application that you have just created.

   c. Program each digit or node individually and set the greetings as desired. Repeat this step for all nodes and digits.

4. Use the System Administrator's mailbox (see **page 863**) to record your custom greetings.

5. Test the application and all of its nodes by calling the application and checking each digit and node.

This section also discusses how to use:

- **Digit Translation** below
- Digit Translation Nodes on **page 886**
- Automatic Fax Detection for CRA Applications on **page 887**

DIGIT TRANSLATION

Applies to UVM voice processors only. Digit translation allows callers to dial a single digit to access a designated extension number, mailbox, or hunt group extension number.

Up to 12 digit translation can be programmed for each digit 0–9, #, and *, plus a Timeout that is used when the caller does not enter a digit. Each digit can lead to a digit translation node (see **page 886**) that has its own digit translation actions. This CRA digit translation creates a tree of programmable digit translation nodes.

To use digit translation, the CRA application must have special programming. Digit translation is programmed by assigning specific "Actions" for each digit and determining what happens if the caller does not enter a digit (Timeout). The Timeout period is determined by the length of the pause at the end of the greeting.

After the application is programmed, the Voice Mail System Administrator's mailbox is used to make a custom audiotex recording that explains the options to callers. For more information, refer to the *MiVoice Office 250 Voice Mail Administrator Guide,*. The custom audiotex recording is then assigned to the CRA application. A caller who listens to the CRA recording can dial the single-digit location number to reach the designated destination. For example, if the hunt group extension for a customer support group is programmed for digit 3, the recording should say something like, "For customer support, press 3."

The system provides a choice between Primary and Secondary Language (see "Multilingual Capability" on **page 888**). The system selects the language to use for each call, as determined by the trunk, phone, and Voice Processing programming. When using a CRA application with digit translation, the individual Voice Processing applications assigned to the digits can be programmed to override the device language and provide prompts in one language only. Using this method, you can program a CRA tree that offers callers a choice between languages.

After you create a digit translation node (see **page 886**), you can assign it to more than one CRA application. This allows entire node hierarchies to be shared or moved without reprogramming. When using a CRA application with digit translation, you can program individual voice processing applications assigned to the digits to override the device language and provide prompts in one language only.

To program digit translations for the CRA:

1. Select Voice Processor– **Devices**.

2. Double-click **Digit Translations**. A list of the digits and their actions and destinations appears.

3. Program the following options:

   • **Name**: The digit that callers press. Timeout represents no digits being entered.

   • **Action**: From the list, select the action for digit. The Actions that can be selected for the digits (*, #, 0–9) and for the Timeout option are as follows:

     • **Company Directory – First Name:** Sends the caller to the directory prompt that asks the caller to enter the first name of the desired party.

     • **Company Directory – Last Name:** Sends the caller to the directory prompt that asks the caller to enter the last name of the desired party.

- **Invalid:** (Not available for Timeout.) The digit will not be used. Callers who press this digit hear a recording that tells them that it is invalid.
- **Subscriber Access:** Sends the caller to the voice mail application that prompts the caller to enter a mailbox number.
- **Transfer To Collected Extension:** (Cannot be used for Timeout, *, #, or 0). To allow callers to dial extension numbers of phones and hunt groups (including off-node devices) that have a mailbox or extension ID, use this Action for digits that correspond to the first digits of extension numbers. For example, if digit 1 is "Transfer To Collected Extension," callers can dial extension numbers that begin with 1. However, if digit 1 is "Transfer to Extension 2000," as described below, callers attempting to dial a phone extension number that begins with 1 will instead be transferred to 2000.

NOTE	If a caller dials an extension number of a mailbox that does not have an associated phone, the call is delivered to the mailbox instead of an extension.

- **Transfer To Extension:** Sends the call to the extension (phone, hunt group, application, or off-node device) that appears in the Transfer Destination field.
- **Transfer To Mailbox:** Sends the call to the designated mailbox.
- **Transfer To Node:** (Not available for Timeout.) Sends the call to a digit translation node that allows access to further digit translation options. See **page 886**. See "Digit Translation Nodes" on **page 886**.
- **Transfer To Operator:** Transfers the call to the programmed Dial-0 Destination.

NOTE	If you select Transfer To Operator but a Dial-0 Destination has not been programmed, a warning message appears, and you must program the Dial-0 Destination before proceeding. For details, see "Dial-0 Destinations" on **page 978**.

- **Hang Up:** This option applies to Timeout only. When applied, the system disconnects from the call if the user does not enter a digit.
- **Transfer Destination**: The actions Transfer To Extension, Transfer To Mailbox, and Transfer To Node each require a Transfer Destination. See the table below to determine the type of destination, if any, to be programmed for the digit translation.

ACTION	TRANSFER DESTINATION
Company Directory - First Name	None
Company Directory - Last Name	None
Invalid	None
Subscriber Access	None
Transfer To Collected Extension	None
Transfer To Extension	Extension number
Transfer To Mailbox	Mailbox number
Transfer To Node	Digit translation node number
Transfer To Operator	None
Hang Up	None

To program the Transfer Destination, use one of the following methods:

Method A

a. Select the current Value, then enter the new value in the text box.

b. Press **ENTER**. A screen appears displaying what is associated with the number entered.

c. Click **OK**. The new number appears in the field.

Method B

a. To change the destination, right-click **Transfer Destination** and choose **Change Transfer Destination**.

b. Select the desired destination type, then click **Next**. A list of fax documents, devices, or digit translation nodes that are present in the system appears.

c. Select the desired destination, then click **Finish**. The selected destination appears in the Transfer Destination field.

- **Override Language**: If this flag is enabled, the Call Routing Announcement application will use the application language (indicated below) for voice prompts. If the flag is disabled, the calling phone or trunk programmed language will be used whenever the application receives a call. By default, this flag is disabled. To enable it, select the check box. To disable it, clear the check box.

- **Language**: If Override Language is enabled, you can select the language from the list. This field can be set to any specific language so that the system can support more than two languages. For more information, see "Language" on **page 242**.

DIGIT TRANSLATION NODES

UVM systems only. A "digit translation node" is a digit translation destination that allows further digit translation options. Unlike extension and mailbox destinations, a digit translation node does not move the call to a specific location. Instead, it offers additional dialing options. A digit translation node can use any of the Actions, including transfers to other digit translation nodes.

The use of digit translation nodes allows the system administrator to create a CRA with several layers to form a digit translation "tree." The tree can be made up of digits with destinations and/ or digit translation nodes with additional translations. There can be up to 200 digit translation nodes in the system and up to 20 per CRA application. An example of a CRA application with a multi-layered tree is shown on the next page.

Once a digit translation node is created, it can be assigned to more than one CRA application. This allows entire digit translation node hierarchies to be shared or moved without reprogramming. A digit translation node can only be deleted if it is not associated with a CRA application.

Digit translation nodes can be programmed to use either the Primary or Secondary Language.

A digit translation node can be used in one or more Call Routing Announcement application. This allows entire digit translation node trees to be shared or moved without reprogramming. There can be up to 200 digit translation nodes in the system.

Each digit translation node can have its own description, username, digit translation programming, and greetings. The digit translation node is assigned as a destination for a digit translation "Transfer To Node" Action. See **page 884**. A digit translation node can only be deleted if it is not associated with a Call Routing Announcement application. Digit translation node programming is very similar to that for a Call Routing Announcement application. The programming fields are described in detail below.

To program a Digit Translation Node:

1. Select Voice Processor – Devices – **Digit Translation Nodes**.

2. Right-click in the right pane, and then click **Create Digit Translation**. The Create Digit Translation Node dialog box appears.

3. Enter the starting extension number and number of extensions, and then click **OK**. The Digit Translation Node appears in the node list.

4. *Optional.* In the Description column, type a description (up to 20 characters). This description is used in the mailbox directory and should be entered in the form "last name, first name" with a comma and space separating the names. Do *not* use Control characters in descriptions.

5. Double-click the new Digit Translation Node.

6. Program the Digit Translations as described on **page 884**.

7. Program the Day and Greetings as described on **page 915**.

AUTOMATIC FAX DETECTION FOR CRA APPLICATIONS

Note: Automatic Fax Detection is not available if the system is running Unified Voice Messaging. Any related fields in Database Programming are marked with a red "X."

With Automatic Fax Detection, CRA applications can be programmed to automatically route incoming fax calls to a specified extension or to an e-mail address. CRA applications can detect fax tones during the greeting and up to timeout. However, the fax tone detection is disabled if the caller performs an action that removes them from the CRA, such as transferring to an extension, transferring to a mailbox, and so forth.

If the feature is enabled, the mailbox or CRA application automatically listens for fax tone as follows:

• A mailbox listens for fax tones during the mailbox greeting and while a message is being recorded. If the mailbox detects fax tones, the call is transferred to the specified extension or e-mail address. If the tones are detected after a recording has started, the call is disconnected.

• Call Routing Applications are able to detect fax tones during the greeting and up to timeout. The fax tone detection is also disabled if the caller performs an action that removes them from the CRA (transferring to an extension, transferring to a mailbox, and so forth).

• If the fax card is busy when an incoming fax call is received, the call is disconnected.

The following two fields control the Automatic Fax Detection feature:

- **Fax Delivery Destination**: This field, if programmed, specifies the extension of the fax machine that will receive incoming faxes.

- **Fax Delivery E-Mail Address**: This field, if programmed, specifies the e-mail address of the account that receives incoming faxes. The fax is converted to a TIFF file and sent to the e-mail address as an attached file. The address can contain up to 127 characters. To view a message, use any TIFF file viewer, such as Imaging for Windows. If you enter an invalid character (: ; " \ | ( ) , < > '), an error tone occurs.

If both fields are programmed with extension numbers, the fax tries delivery at the e-mail address. If all fax ports are unavailable, the fax goes to the extension specified in the Fax Delivery Destination field.

MULTILINGUAL CAPABILITY

UVM systems only. The multilingual capability provides a choice between American English, British English, Spanish, Canadian French, and Japanese prompts and displays. The system selects the language to use for each call, as determined by the trunk, phone, and Voice Processing programming.

If Secondary Language language prompts are not installed and a user attempts to access them, Primary Language prompts are delivered instead.

NOTES	Multilingual capability requires the Multilingual standard feature. This prevents users from unintentionally changing the language in an all-English system.
	52xx/53xx-series IP phones do not support the Japanese language.

For complete information about multilingual capability, see "Multilingual Capability" on .

MULTILINGUAL VOICE PROCESSING

When a Voice Processing application receives a call from a phone or trunk, the system tells the application which language is programmed for that device. For example:

- If a trunk programmed for the Primary Language rings in to an application, the voice processor plays the Primary Language prompts.

- If a phone programmed for the Secondary Language calls Voice Mail, the phone user hears Secondary Language prompts.

- If a phone programmed for the Primary Language receives a call on a trunk that is programmed for the Secondary Language, and then forwards the call to Voice Mail, the caller hears Secondary Language voice prompts.

MULTILINGUAL CRAS WITH DIGIT TRANSLATION

When using a CRA application with digit translation, the individual Voice Processing applications assigned to the digits can be programmed to override the device language and provide prompts in one language only. Using this method, you can program a CRA tree that offers callers a choice between languages. For example, in a system that uses English and Japanese:

- The CRA application could have a greeting that says, "Thank you for calling. For English prompts, press 1. NIHONGO WA, 2 WO OSHITE KUDASAI (for Japanese, press 2)."

- The digit translation for digit 1 would lead to a digit translation node that overrides the calling device programming and uses only English prompts.

- The digit translation for digit 2 would lead to a digit translation node that overrides the calling device programming and uses only Japanese prompts.

In the example above, the digit translations could be digit translation nodes that lead to various other English-only or Japanese-only applications. Or, the first level can give more choices, such as English Voice Mail or automated attendant and Japanese Voice Mail and automated attendant. Either way, the individual applications or digit translation nodes can be programmed to play only one language or they can use the device language, as needed.

MULTILINGUAL FEATURES ON NETWORK OPERATION

Nodes do not need to match languages because the Voice Mail system installs any existing language prompt, not just the languages designated as Primary or Secondary. Therefore, for calls across network nodes, if the language does not match between the two nodes, the called Voice Mail system searches for the designated language in its system in the following order: Primary language, Secondary language, American English, British English, Japanese, Canadian French, and Spanish and uses the designated language. If the designated language does not exist in the Voice Mail system, the system uses the default Primary language instead.

MESSAGE NOTIFICATION/RETRIEVAL

This section contains the following information:

- **Feature Description** below

- Programming MNR Classes of Service on **page 891**

The Message Notification/Retrieval (MNR) application is required for message notification, Remote Messaging, and Fax-On-Demand. You must create one (only) MNR application for the system to allow voice mail message notification and quick mailbox access.

Accept AMIS Calls Flag

This flag determines whether the CRA accepts incoming AMIS (Audio Messaging Interchange Specification) calls. If this flag is enabled, when the CRA answers an incoming CO call, in addition to detecting the normal DTMF tones (0–9, *, and #), it also looks for the DTMF tone "C." If it detects a C tone, the application begins processing the AMIS call. If this flag is disabled, the application does not look for the DTMF tone C. In this case, if the application answers an incoming AMIS call, it treats the call as a normal call (which means it will eventually time out). The default value for this field is Disabled. To enable it, select the check box to place a check in it. To disable it again, select the check box again to remove the check.

NOTES	Call Routing Announcement applications programmed for AMIS cannot have greetings programmed because the AMIS protocol causes the destination system to disconnect if voice is detected. UVM does not support AMIS networking.

Because the MNR application places outgoing calls for remote notification and Fax-On-Demand, it can have toll restriction classes of service. Fax-On-Demand is *not* available in UVM.

FEATURE DESCRIPTION

Two application types are combined to provide the Voice Mail feature: Voice Mail and Message Notification/Retrieval.

- **Voice Mail:** This application handles all calls that are directed to Voice Mail (other than the Message Notification/Retrieval application) placed by subscribers and non-subscribers. Callers hear the main company greeting, followed by a menu of available options. Phones can forward or transfer calls directly to their mailbox using this application's extension number. This application can also be the message center for the subscribers' phones.
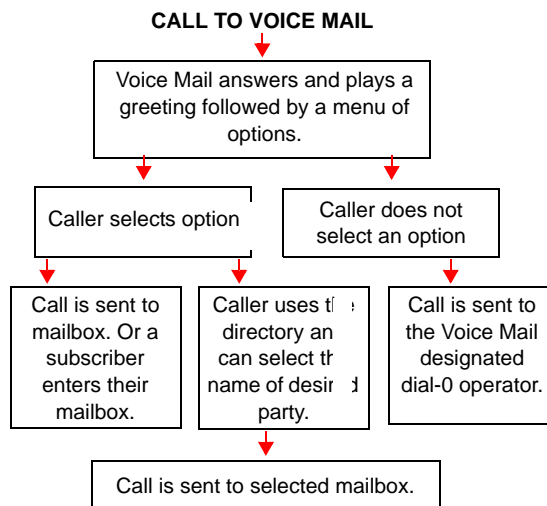
CALL TO VOICE MAIL

Voice Mail answers and plays a greeting followed by a menu of options.

Caller selects option

Caller does not select an option

Call is sent to mailbox. Or a subscriber enters their mailbox.

Caller uses the directory and can select the name of desired party.

Call is sent to the Voice Mail designated dial-0 operator.

Call is sent to selected mailbox.

- **Message Notification/Retrieval:** Only one Message Notification/Retrieval application may be created for the system. When phone users respond to message indications left by Voice Mail, they reach the Message notification/Retrieval application and are prompted only for their password.

Phone users and outside callers gain access to the Voice Mail application by:

- Dialing the telephone number or extension number assigned to the Voice Mail application.

- Being transferred to the Voice Mail by the automated attendant.

- Being forwarded to the Voice Mail when a called phone is unavailable (through the Call Forwarding or System Forwarding features).

- Responding to message waiting indications.

MESSAGE IS LEFT IN A MAILBOX

↓

Voice mail leaves a message waiting indication at the phone designated as the message notification phone for the mailbox.

↓

Phone user responds to message indication (presses MSG button or enters feature code).

↓

INTERCOM CALL IS PLACED TO MESSAGE NOTIFICATION/RETRIEVAL APPLICATION (Alternate Message Source)

↓

Phone user is prompted for the password to his mailbox.

Callers hear the Voice Mail company greeting and recorded instructions that tell them what to do next. Users simply listen to the prompts and press the keypad button that corresponds to the desired choice. If the user does not respond immediately, a second set of prompts is played. Most prompts are interruptible, and users can press the desired button at any time during the prompt. The prompt then stops, and the system acts on the requested choice.

PROGRAMMING MNR CLASSES OF SERVICE

To program the toll restrictions, double-click **Classes of Service**; you have the option of choosing **Day** or **Night**. Double-click the desired time period to view a list of current classes of service.

To add a class of service:

1. Right-click in the window and select **Add To List**.

2. A window appears that allows you to select the types of toll restrictions you want to program. Select **ARS Only & Deny Area/Office** and/or **Classes of Service**, then click **Next**.

3. Another window appears that shows you a list of available classes of service. You can view them in a list by selecting the List button or view details by selecting the Details button.

4. Select the desired class(es) of service, then select **Add Items**. The selected classes of service appear in the list. Click **Finish** to exit.

5. *(U.S. only) If you selected Deny Area/Office class of service,* the application must also be assigned to a user group. To change the user group, right-click **User Group** and select **Change User Group**. In the first window that appears, select **User Groups**, then click **Next**. In the next window, select the desired user group, then click **Finish** to exit and save the change.

To delete a class of service:

1. Select the item(s).

2. Right-click and select **Remove Selected Items**. To select a series of items, hold down **SHIFT** while selecting the first and last item in the range. To select two or more that are not consecutive, hold down **CTRL** while selecting the desired items.

RECORD-A-CALL

This section describes the Record-A-Call feature using UVM and MiCollab Unified Messaging. This section contains the following information:

*   **Feature Description** below

*   UVM Record-A-Call on **page 894**

*   MiCollab Unified Messaging Record-A-Call on **page 895**

FEATURE DESCRIPTION

If your system is programmed with a Record-A-Call application, the phones can be programmed to use the Record-A-Call feature. It allows users to enter a feature code whenever they want to record an ongoing call in their designated Record-A-Call mailbox. Users can retrieve the recorded messages later, just as they would any other mailbox messages.

NOTES	The Record-A-Call standard feature is required. Call monitoring may be illegal in some locations. It is the responsibility of the customer to ensure that the use of this feature complies with local laws.

When a user requests the Record-A-Call feature, the system establishes a conference call with the current call parties and a mailbox. If no conference circuits are available when the user requests Record-A-Call, or if there are already four parties on the call, the user hears reorder tones and cannot use the feature.

For feature usage instructions, refer to the applicable phone user guide.

You can program a DB Programming flag that enables Pre-Record-A-Call messaging capability. This allows the MiVoice Office 250 user to play a Voice Processor message informing the parties on the call that their conversation is about to be recorded with the Record-A-Call feature.

The Record-A-Call mailbox records the call as a voice mail message. All parties will be included in the recording. If desired, the Record-A-Call mailbox can be programmed to play a message announcing that the Record-A-Call feature is in progress. Separate messages can be recorded for day and night modes.

To signal to the other parties that the Record-A-Call feature is in use, a system-wide Record-A-Call Tone flag can be enabled during system programming. If the flag is disabled, there will be no alerting tone. If enabled, the beep will occur periodically throughout the call. The Record-A-Call Tone Interval timer determines how often this tone is generated. If the timer is set to 0, the tone is generated only when the feature is first activated.

You can program a phone to use this feature by one of the following two methods:

*   The phone can be programmed to use its personal mailbox or another mailbox, as the assigned Record-A-Call mailbox. Only this assigned mailbox can be selected. (This is the default programming for all phones.)

*   The phone can be programmed with a default mailbox but with the option of selecting a different mailbox. If the user chooses to not enter a mailbox number, the system automatically selects the default mailbox.

In a network setting, the Record-A-Call destination does not have to be on the same node as the phone, but the voice processor must be programmed with a mailbox for that phone. When the requesting phone user hangs up, all parties are automatically disconnected. If all parties on the call hang up except the requesting phone user, an intercom call will remain connected between the requesting phone and the Record-A-Call application. This allows the user to make additional comments before ending the recording.

When the phone user turns off Record-A-Call or ends the call being recorded, the system delivers the message to the mailbox. The associated phone will receive message waiting indications as usual.

Record-A-Call can be used during Agent Help and three-party conference calls. In addition, a monitored phone can initiate a Record-A-Call session without terminating call monitoring. Hunt group supervisors, however, cannot initiate a Record-A-Call while monitoring a phone, and a phone using Record-A-Call cannot be monitored. In other words, to use Record-A-Call and Station Monitor features simultaneously, the supervisor must first be monitoring the phone, and the phone, not the supervisor, must initiate the recording.

The Record-A-Call feature code can be assigned to a user-programmable feature button. If the feature button has a lamp, it flashes while the Record-A-Call conference is being set up and is lit while the feature is active.

As with any other conference call, any inside party involved in a Record-A-Call conference can use the Transfer, Agent Help, or other features. However, if any inside party has enhanced speakerphone enabled, the enhanced mode will be disabled when the conference begins and must be re-enabled if desired.

> ### NOTICE
> Pressing Hold during a record-a-call will terminate the record-a-call.

UVM RECORD-A-CALL

This feature allows users to record an ongoing call and place it in a voice mailbox. When a user enters the Record-A-Call feature code, the system places a call to the phone assigned Record-A-Call application using the phone Record-A-Call mailbox. When the application answers, the system sets up a conference call. If programmed, the mailbox plays a greeting to indicate that the recording is in progress. There can be separate greetings for day and night modes.

If the external Voice Processing system has a Record-A-Call application, users can enter a feature code to record ongoing calls in their designated Record-A-Call mailboxes. Users can retrieve the recorded messages later, just as they would any other mailbox message.

NOTES | This feature requires the Record-A-Call software license. Call monitoring may be illegal in some locations. It is the responsibility of the customer to ensure that the use of this feature complies with local laws.

The Record-A-Call mailbox records the call as a Voice Mail message. All parties are included in the recording. If desired, the Record-A-Call mailbox can be programmed to play a message announcing that the Record-A-Call feature is in progress. There can be separate messages for day and night modes. A Record-A-Call tone can be programmed to alert callers at the beginning of the recording. Also, it can be programmed to beep periodically throughout the recording. Record-A-Call tone is enabled at default.

There are two ways a phone can be programmed to use this feature:

- The phone can use its personal mailbox, or any other mailbox, as the assigned Record-A-Call mailbox. No other mailbox can be selected. (This is the default programming for all phones.)

- The phone can be programmed with a default mailbox but with the option of selecting a different mailbox. If the user chooses to not enter a mailbox number, the system automatically selects the default mailbox.

When the requesting phone user hangs up, all parties are automatically disconnected. If all parties on the call hang up, except the requesting phone user, an intercom call remains connected between the requesting phone and the Record-A-Call application. This allows the user to make additional comments before ending the recording.

When the phone user turns off Record-A-Call or ends the call being recorded, Voice Processing delivers the message to the mailbox. The phone associated with the mailbox receives message waiting indications as usual. Refer to the appropriate phone user guide for message retrieval instructions.

The system administrator can set a maximum length for Record-A-Call messages. The Record-A-Call Maximum Message Length timer can be set at 0–600 minutes. A 0 setting allows messages of any length, limited only by the available disk space. The default is 30 minutes.

In a network setting, the Record-A-Call destination does not have to be on the same node as the phone, but Voice Processing must have a mailbox for that phone.

To enable Record-A-Call:

1. Select Voice Processor – Devices – **Record-A-Call**.

2. Right-click in the right pane, and then click **Create Record-A-Call**. The Create Record-A-Call dialog box appears.

3. Enter the starting extension number and number of extensions, and then click **OK**. The Record-A-Call extension appears in the node list.

4. *Optional.* In the Description column, type a description (up to 20 characters). This description is used in the mailbox directory and should be entered in the form "last name, first name" with a comma and space separating the names. Do *not* use Control characters in descriptions.

5. Click out of the field or press **ENTER** to save the change.

MICOLLAB UNIFIED MESSAGING RECORD-A-CALL

This section contains the following information:

- **Functional Behavior** below

- Programming MiCollab Unified Messaging RAC on **page 900**

- Troubleshooting MiCollab Unified Messaging Record-A-Call on **page 907**

- Diagnostics on **page 908**

NOTICE

Call monitoring may be illegal in some locations. It is the responsibility of the customer to ensure that the use of this feature complies with local laws.

Using the MiCollab Unified Messaging RAC feature, MiVoice Office 250 users can record a phone conversation while it is in progress. After the recording has ended, the RAC application delivers the recorded call to the user's mailbox.

The RAC feature also provides a recorded memo option. Using this option, users can call the RAC application and record a memo message. Like recorded calls, recorded memos are saved in the user's mailbox.

Users can access recorded calls and memos using one of the following MiCollab Unified Messaging interfaces:

- Mailbox (default voice mailbox or Record-a-Call mailbox)
- Mitel Applications Suite (MAS) End-User Portal

In general, the user can play, reply to, forward, and delete call and memo recordings created by the RAC feature. Refer to one of the following for detailed information and instructions for recorded call options:

- MiCollab Unified Messaging User Guide
- MAS End-User Portal Help

Mitel documentation is available on the eDocs Web site (http://edocs.mitel.com/).

FUNCTIONAL BEHAVIOR

This section provides functional details for MiCollab Unified Messaging RAC on the MiVoice Office 250.

Licensing

RAC is a licensed feature on MiCollab Unified Messaging. MiCollab Unified Messaging customers must purchase the RAC license, and administrators must configure the license for use on the system. There are no licensing requirements for MiCollab Unified Messaging RAC on the MiVoice Office 250.

Resources and Capacities

There is no set capacity for maximum number of simultaneous RAC sessions because RAC availability depends on the number of:

- existing conference calls on the originating node.
- existing conference calls for all networked nodes.
- existing MiCollab Unified Messaging memo Record-a-Call sessions.
- MiCollab Unified Messaging voice mail ports available.
- IP resources available in the system.

Similar to UVM, the MiCollab Unified Messaging RAC feature requires the following resources:

- **MiVoice Office 250 Conference Resources**: Because call processing creates a conference call to MiCollab Unified Messaging to establish the RAC session, the number of available conference resources on the system/network at a given time affect the RAC feature. If a user requests MiCollab Unified Messaging RAC and there are no conference resources available, the RAC feature fails. This also applies to UVM.

    NOTE Memo recordings do not require conference resources.

- **MiCollab Unified Messaging Voice Mail Ports**: Each MiCollab Unified Messaging RAC request requires at least one available MiCollab Unified Messaging voice mail port so that call processing can establish a call to the MiCollab Unified Messaging system and connect that call with the RAC conference call. A voice mail port is required for both recorded calls and recorded memos. If a user requests MiCollab Unified Messaging RAC and there are no voice mail ports available on MiCollab Unified Messaging, the RAC feature fails.

- **MiVoice Office 250 IP Resources**: An IP resource is a quantity of processing power that digital signal processors (DSPs) on the system use for call processing functions. In terms of processing power consumed, the "cost" of a DSP function varies, depending on whether the DSP resides on the Processor Module (PM-1) or on the Processor Expansion Card (PEC-1) in the Base Server. Because the MiCollab Unified Messaging RAC feature requires system IP resources, verify that IP resources are properly allocated on the system. If required, use the Resource Reservation Tool in DB Programming.

Table 156 provides the MiVoice Office 250 capacities for the resources required by MiCollab Unified Messaging RAC.

Table 156: MiVoice Office 250 Resources and Capacities Affected by MiCollab Unified Messaging RAC

RESOURCE	MAXIMUM CAPACITY
Conference	
• Total per system	20
• Parties per conference	4
• Simultaneous 4-party conferences	5
MiCollab Unified Messaging Voice Mail Ports	32

NOTE System IP resource capacities vary based on several factors such as server model, voice channels used, and IP resource requirements for devices and features. Therefore, IP resource capacities are not included in the table.

Call Processing Response to RAC Requests

Depending on the request from the user (RAC or memo), MiVoice Office 250 Call Processing responds as follows:

- **RAC requests**: When users initiate the RAC feature, the MiVoice Office 250 establishes a conference call with the user, the other parties on the call, and the MiCollab Unified Messaging system. The conference call provides MiCollab Unified Messaging with the mailbox extension for the user who initiated the RAC feature.

- **Memo requests**: When the user dials the RAC extension, MiVoice Office 250 Call Processing initiates an intercom call to MiCollab Unified Messaging requesting a RAC session. This call provides MiCollab Unified Messaging with the user's mailbox extension so that the recorded memo can be saved in the user's mailbox.

Interoperability with Other Features

The following information explains how MiCollab Unified Messaging RAC interoperates with other MiVoice Office 250 features:

- **Conference**: The MiCollab Unified Messaging RAC feature interacts with the conference feature as follows:

  - *Recording calls*: MiVoice Office 250 call processing sets up a conference call to MiCollab Unified Messaging to create a RAC session. Conference calls initiated by a RAC session require adequate conferencing resources and function like regular conference calls. If there are no conference circuits available on the system, or if there are already four parties in the existing call, the RAC feature fails.

    Typically, the user who initiated the RAC feature ends the conference by hanging up. However, if the RAC session included the user and two other parties, the remaining parties stay connected on the conference call until they hang up. If all of the other conference parties hang up before the RAC user, the user remains connected to the RAC session. This allows the user to add comments to the end of the recording.

  - *Recording memos:* Unlike recorded calls, call processing does not create a conference call for memo recordings. Therefore, memo recordings do not consume conference resources in the system. The maximum number of memo recordings at a given time is limited only by the available number of MiCollab Unified Messaging voice mail ports and IP resources on the system.

- **Hold**: When the user places a recorded call on hold, the recording is terminated. To resume recording, the user must remove the call from hold and then start the RAC feature again. The hold feature is not supported for memo recordings.

- **Camp-on**: RAC calls do not have the ability to camp-on.

  RAC memos camp-on when the **Camp-Ons Allowed** option is enabled (System – Devices and Feature Codes – SIP Peers – SIP Voice Mails – P90001 – Configuration) and one of the following occurs:

  - The system does not have any available IP resources.
  - The system does not have any available voice mail ports.
  - The SIP peer voice mail is in the **Out-of-Service** or **Out-of-Service-Maintenance** state. For complete information about SIP peer voice mails, see "SIP Voice Mails" on **page 711**.

- **Audio Diagnostics**: If the RAC feature is programmed for the phone, the user can record the call while using Audio Diagnostics feature. This option is presented after the user accesses the Audio Diagnostics feature (feature code 320).

- **Peer-to-Peer Media**: The RAC feature is not available on calls using Peer-to-Peer Media.

- **System Open Architecture Interface (OAI)**: For OAI purposes, the MiCollab Unified Messaging RAC feature functions like the RAC feature supported by UVM. There are no changes required on the OAI applications to use the MiCollab Unified Messaging RAC feature.

  The following OAI applications are able to use SIP Record-a-Call feature:

  - Attendant Console
  - Customer Service Manager (CSM)
  - UC 5000

Feature Limitations

The RAC feature has the following limitations:

- The RAC feature is not available for devices that are using peer-to-peer media.
- You cannot place a recorded memo call on hold.
- The RAC feature code (385) is not available for phantom extensions.
- Hunt group supervisors cannot initiate the RAC feature if they are currently using the Station Monitor feature to monitor a member of the hunt group.

Functional Differences from UVM

The following list provides the functional differences between the MiCollab Unified Messaging and Unified Voice Messaging (UVM) RAC. Other than the differences indicated below, the MiCollab Unified Messaging RAC feature functions the same as the UVM RAC feature.

Functional differences between MiCollab Unified Messaging RAC and UVM RAC include:

- **Message Waiting Indication (MWI)**: UVM recorded calls and memos are stored as new messages in users' mailboxes and therefore trigger the MWI on the user's phone. In contrast, MiCollab Unified Messaging treats recorded calls as recorded conversation so they do not trigger the MWI on the user's phone.
- **Pre-recorded messages**: UVM RAC applications include an option to play a recorded message at the beginning of the RAC session. This option is configured under System – Flags – **Play Pre-Record-a-Call Message**. The MiCollab Unified Messaging RAC application does not include the recorded message option. Therefore, when you configure the system to use MiCollab Unified Messaging, this flag appears with a red **X**.
- **Recording length**: For UVM you can specify the RAC length limit in DB Programming under Voice Processor – Timers and Limits – Timers and Limits – **Record-A-Call Max Message Length**. The range for this timer is 1-120 minutes and the default is 30 minutes. MiCollab Unified Messaging does not include a configuration option to specify the length limit on RAC recordings. Instead, MiCollab Unified Messaging provides a fixed, non-configurable limit of 60 minutes for the RAC recording length.
- **Silence during a recording**: For UVM recorded calls and memos, if the recording encounters silence, the recording continues for the duration of the call. For MiCollab Unified Messaging recorded calls and memos, if the recording encounters silence for 60 seconds, the recording is automatically terminated. The 60-second Silence Suppression timer is a fixed, non-configurable timer for MiCollab Unified Messaging RAC.

Timeouts

If the MiVoice Office 250 SIP peer voice mail is in service but call processing cannot connect to MiCollab Unified Messaging for some reason, then a pending RAC request will time out. There can be a number of reasons why call processing cannot connect to MiCollab Unified Messaging (for example, network problems, configuration errors, MiCollab Unified Messaging is currently offline, and so forth).

There are two timers that determine timeout conditions:

- **Call Processing Timeout**: The call processing timer is set to 15 seconds and is not configurable in DB Programming. If the user requests RAC and the system cannot complete the request in 15 seconds or less, the **CALL CANNOT BE COMPLETED** message is displayed.

- **SIP Message Timeout**: This timer is configured in DB Programming. The field is available in Online Monitor Mode (OLM) under System\Devices and Feature Codes\SIP Peers\SIP Voice Mails\<*P9xxx*>\Configuration**Timers**. By default, this timer is set to 16000 milliseconds (16 seconds). If the SIP Message Timeout value is less than the Call Processing Timeout value, and the threshold is reached, the **RECORD-A-CALL REJECTED** message is displayed.

NOTE	Do *not* use OLM mode unless you are instructed to do so by Mitel technical support personnel.

When a memo request times out (SIP or CP timer), the **DESTINATION IS NOT RESPONDING** message is displayed.

PROGRAMMING MICOLLAB UNIFIED MESSAGING RAC

To provide MiCollab Unified Messaging RAC functionality on the MiVoice Office 250, you must complete the associated configuration for both the MiVoice Office 250 and MiCollab Unified Messaging.

The following diagram illustrates a typical configuration for the RAC application along with the other SIP peer voice mail applications on the MiVoice Office 250. Note that there is no mailbox application or line group configured on the MiCollab Unified Messaging system that corresponds to the SIP RAC application. The Voice Mail Line Group application on MiCollab Unified Messaging handles the RAC functionality.

Therefore, when you program the RAC application in DB Programming:

- Assign the RAC application a unique extension.

- Program the RAC pilot number to be the same extension as the Voice Mail application extension for the SIP peer voice mail. The Voice Mail application extension matches the pilot number of the Voice Mail Line Group on the MiCollab Unified Messaging system.

For complete information about SIP peer voice mails, see "SIP Voice Mails" on **page 711**.
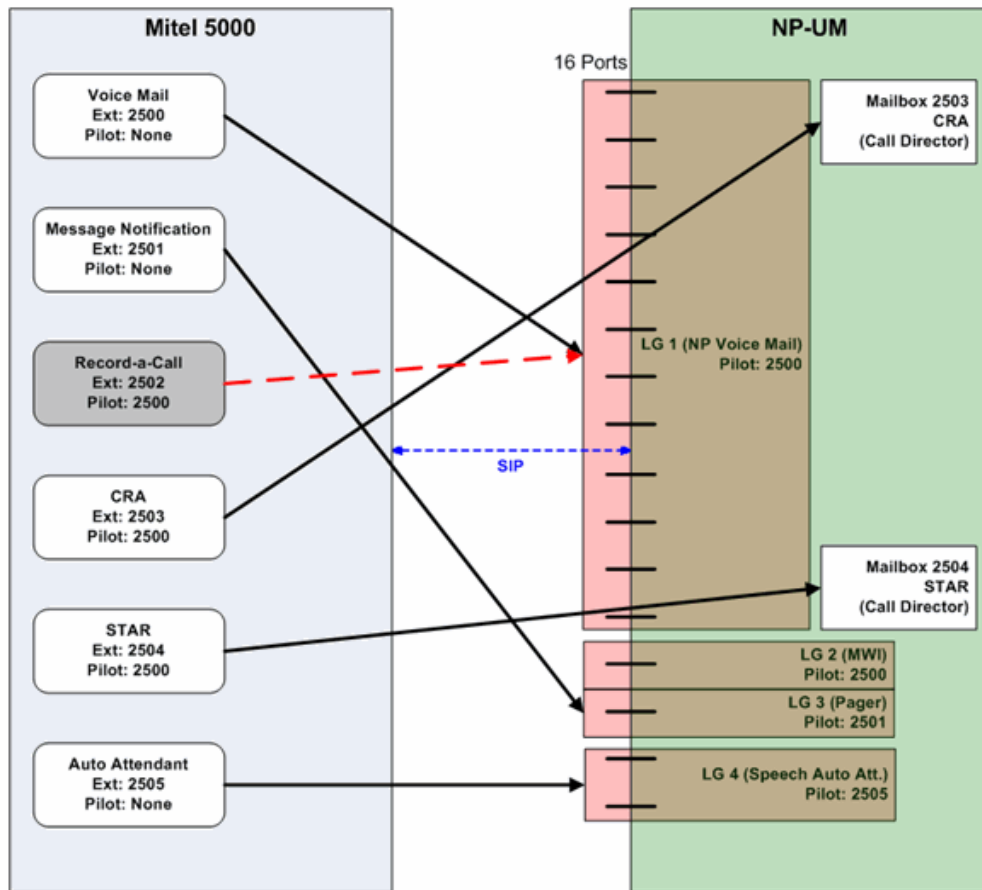
MiVoice Office 250 Configuration

Before you can configure the MiCollab Unified Messaging RAC application in DB Programming, you must configure MiCollab Unified Messaging as the messaging platform for the MiVoice Office 250. Refer to the *MiVoice Office 250 and MiCollab Unified Messaging Integration Guide*, for integration information and initial configuration instructions.

Be aware of the following fields and options in DB Programming that affect the MiCollab Unified Messaging RAC feature:

- **SIP Voice Mail Operating State**: The SIP Voice Mail must be placed in the **In-Service** operational state for the RAC feature to work. If the SIP peer voice mail is in **Out-of-Service** or **Out-of-Service-Maintenance** state, the following conditions occur:

    - All requests for RAC fail. The user hears a tone, and the **CALL CANNOT BE COM-PLETED** message appears on the user's display.

    - Memo recordings may fail, depending on how the SIP Voice Mail **Camp-Ons Allowed** option is configured. The Camp-Ons Allowed option is located under System\Devices and Feature Codes\SIP Peers\SIP Voice Mails\<*P9xxx*>\Configuration. Options in-clude **Yes** (enabled), and **No** (disabled):

        - If the **Camp-Ons Allowed** flag is set to **Yes**, memo requests will camp-on and the <*RAC*> **IS UNPLUGGED**, **WAITING FOR** <*RAC*> series of messages appears on the user's display.

- If the **Camp-Ons Allowed** flag is set to **No**, the call does not camp-on and the **<*RAC*> IS UNAVAILABLE** message appears.

NOTE	When programming the RAC feature, temporarily place the SIP peer voice mail in the **Out-of-Service- Maintenance** operating state. When you are finished programming, place the SIP Voice Mail in the **In-Service** operating state.

See **pages 902** and **904** for programming instructions.

- **System Flags**: The following options apply to the MiCollab Unified Messaging RAC feature:

  - **Record-a-Call Tone**: Determines whether the parties on a call will hear a tone when the MiCollab Unified Messaging RAC feature is in use. By default, this flag is set to **Yes**, which means that the parties will hear the RAC tone. The **Record-a Call Tone Interval** timer determines how often the tone is played during the recording.

  - **Record-a-Call Display**: Enables or disables RAC displays for the system. By default, this flag is set to **Yes**, which means all users with display phones will see RAC display messages.

  See **page 904** for programming instructions.

- **Mailboxes:** Determine the mailbox configuration you want to use for RAC, and then program the mailboxes accordingly. Options include:

  - **RAC mailbox:** If you want users to have the option of saving RAC messages in a separate RAC mailbox, you will need to program those additional mailboxes in DB Programming.

  - **Default voice mailbox**: If you want users to save RAC messages in their default voice mailbox (along with voice mail messages), you do not need to configure extra mailboxes.

  For every mailbox you create in MiVoice Office 250 DB Programming, you need to create a corresponding mailbox in MiCollab Unified Messaging.

- **Record-a-Call Tone Interval**: Determines how often the Record-a-Call Tone is played during a recording. This timer has a range of 0-255 seconds. By default, the timer is set to **0**, which means the tone is played at the beginning of the recording only. See see **page 904** for programming instructions.

For additional information and related programming instructions, refer to the following MiVoice Office 250 documentation:

- For programming instructions for SIP voice mails (MiCollab Unified Messaging on the MiVoice Office 250), see "SIP Voice Mails" on **page 711**.

- For SIP peer feature information, see "Service Provider SIP Trunks and SIP Trunk Groups" on **page 691**.

- For SIP peer licensing information, refer to the *MiVoice Office 250 Installation Manual* .

To configure MiCollab Unified Messaging RAC in MiVoice Office 250 DB Programming:

1. Take the SIP peer voice mail that you created for MiCollab Unified Messaging out of service for maintenance.

    a. Select System – Devices and Feature Codes – SIP Peers – SIP Voice Mails – <P92001> – Configuration – **Operating State**.

      b. Select **Out of Service-Maintenance** from the list box, and then press **ENTER** or click out of the options to save the setting.

2. Create the SIP RAC application.

      a. Select System – Devices and Feature Codes – SIP Peers – SIP Voice Mails – *<P92xxx>* – **Applications**.

      b. Right-click and select **Create Record-A-Call**.

      c. Select the extension for the application, and then click **OK**.

      d. Program a **Description** and **Username** for the RAC application.

3. Double-click the RAC extension you created in **step 2** to access the RAC fields.

4. Configure the following SIP RAC application fields:

- **Class of Service**: Class of Service (COS) is used for toll restriction, which prevents system users from placing outgoing calls. The COS fields include:
  - Day and Night Lists
  - Dialing Patterns
  - Allow Dialing Patterns

- **SIP Voice Mail Pilot**: Indicates the SIP Voice Mail pilot number. Configure the SIP Voice Mail Pilot number to be the SIP Voice Mail application extension.

- **Transfer Recall Destination**: Indicates the phone, hunt group, or application that receives any calls that recall after being transferred by the application.

- **Attendant**: Indicates the phone, application, or hunt group that serves as the attendant for the application.

- **Music-On-Hold Source Fields**: RAC applications can be programmed to determine the music source a caller hears based on the device and the status of the call. Music-On-Hold fields include:
  - Audio for Calls Camped onto this Device
  - Audio for Calls Holding for this Device
  - Audio for Calls Ringing this Device

- **Calling Party Name**: This field is similar to the Calling Party Number field. It is used only for ISDN calls to the public network (non-private networking). If this field is set, the system may use this information for the outgoing ISDN setup request message.

- **Calling Party Number**: Each application can be programmed to send an identifying number when a call is placed.

- **Prompt for Mailbox on Transfer**: Indicates whether a transfer announcement call should prompt a transferring party for a mailbox number. This option is set to **No** by default.

- **Propagate Original Caller ID on Transfer**: Allows the system to pass the caller ID name or number on an outgoing ISDN call if the call has not been answered by the system (extension, voice mail, hunt groups, or OAI application) or for transfer announcement calls. This field is intended for customers that want to route incoming calls from the MiVoice Office 250 back to the PSTN via ISDN lines. By default this flag is set to **No**.

- **Validate Mailbox on Transfer**: Indicates whether the system validates the collected mailbox number on a transfer. This option is set to **No** by default.

NOTE	Refer to the *MiVoice Office 250 DB Programming Help* for field descriptions and additional programming information.

5. Select System – **Flags**, and configure the following options for the SIP RAC application:

- **Record-a-Call Display**: Enables or disables RAC displays for the system. By default, this flag is set to **Yes**, which means all users with display phones will see RAC display messages (see **page 905**).

- **Record-a-Call Tone**: Determines whether the parties on a call will hear a tone when the MiCollab Unified Messaging RAC feature is in use. By default, this flag is set to **Yes**, which means that the parties will hear the RAC tone. The **Record-a Call Tone Interval** timer (see **step 6**) determines how often the tone is played during the recording.

6. If you enabled the **Record-a-Call Tone** (see **step 5**), configure the **Record-a-Call Tone Interval** (System – **Timers and Limits**). This timer determines how often the Record-a-Call Tone is played during a recording. The range for this timer is 0-255 seconds. By default, it is set to 0, which means the tone is played at the beginning of the recording only.

7. Select System – Devices and Features Codes – Phones – *<phone>*, and then program the following **Record-A-Call** fields for MiVoice Office 250 phones:

- **Mailbox**: Options include:
  - SIP Associated Mailbox
  - SIP Non-Associated Mailbox
  - This Station's Associated Mailbox

- **Mailbox User-Keyed Extension**: Select one of the following:
  - **No**: The system does not prompt the user to enter a RAC mailbox number for RAC requests. The system automatically sends RAC messages to the mailbox configured in the Mailbox field.
  - **Yes:** The system prompts the user for a RAC mailbox when initiating RAC requests.

- **Application**: Select the SIP RAC application you created in **step 2**.

NOTE	Refer to the *MiVoice Office 250 DB Programming Help* for field descriptions and additional programming information.

8. Repeat **step 7** for all of the phones that you want to program for MiCollab Unified Messaging RAC.

9. Place the SIP peer voice mail in service.

   a. Select System – Devices and Feature Codes – SIP Peers – SIP Voice Mails – *<P92001>* – Configuration – **Operating State**.

   b. Select **In-Service** from the list box, and then press **ENTER** or click out of the options to save the setting.

MiCollab Unified Messaging Configuration

You must program several fields in the MAS interface to make the feature available to MiVoice Office 250 users.

Be aware of the following non-configurable timers and limits for MiCollab Unified Messaging:

- **RAC Recording Length**: MiCollab Unified Messaging provides a fixed, non-configurable limit of 60 minutes for the RAC recording length. The recording and associated conference call is automatically ended after 60 minutes.

- **RAC Silence Suppression**: MiCollab Unified Messaging provides a fixed, non-configurable silence suppression limit of 60 seconds for RAC, meaning, that the RAC application will automatically hang up if it detects 60 seconds of silence during a RAC session. When the RAC session is ended, the user hears a beep and the **RECORD-A-CALL TERMINATED** message appears on the phone display.

Listed below is an overview of the configuration required on MiCollab Unified Messaging for the RAC feature.

To configure RAC on MiCollab Unified Messaging:

1. Obtain the required licensing.

2. If necessary, install the **NPMRecordACall** software blade.

3. Configure MiCollab Unified Messaging to work with the MiVoice Office 250. For every mailbox you created in MiVoice Office 250 DB Programming, you need to create a corresponding mailbox in MiCollab Unified Messaging.

4. Configure the RAC Feature Class of Service (FCOS) for mailboxes.

For detailed information about MiCollab Unified Messaging, refer to the following MiCollab Unified Messaging documentation:

- For Engineering information such as system and networking requirements, system capacities, and deployment scenarios, refer to the *MiCollab Unified Messaging Engineering Guidelines*.

- For technical information including software and hardware specifications, installation and software upgrades, and programming information, refer to the *MiCollab Unified Messaging System Administration Help* and the *MiCollab Unified Messaging Technician's Handbook*.

- For information and instructions for MiCollab Unified Messaging administrator tasks such as managing mailboxes, generating reports, and system maintenance tasks, refer to the *MiCollab Unified Messaging Web Console Help*.

Record-a-Call Display Messages

The **Record-a-Call Display** flag in DB Programming (System – Flags) determines whether Record-a-Call display messages appear for phones requesting the RAC feature (see **page 904**). When this flag is set to **Yes**, the display messages in **Table 157** are enabled for users with display phones.

Table 157:   MiCollab Unified Messaging RAC Display Messages

DISPLAY MESSAGE	INDICATES
REQUESTING RECORD-A-CALL	The user has initiated the RAC feature.
RECORD-A-CALL IN PROGRESS	A RAC session is in progress.
RECORD-A-CALL TERMINATED	The RAC session has been terminated due to one of the following: • The mailbox is not configured. • The mailbox is full. • The mailbox does not have the RAC Feature Class of Service (FCOS). • The RAC session was terminated by the user.
RECORD-A-CALL REJECTED	The RAC request was rejected due to one of the following: • All the voice mail ports are in use and the RAC feature is temporarily unavailable. • All of the system IP resources are in use and the RAC feature is temporarily unavailable. • All of the system conference resources are in use and the RAC feature is temporarily unavailable. • The RAC request timed out due to the SIP Message Timeout timer.
IC TO *<RAC>*[1]	The user has initiated a memo recording.
CALL CANNOT BE COMPLETED	the Record-a-Call feature has failed due to one of the following: • The SIP peer voice mail is in the **Out-of-Service** or **Out-of-Service-Maintenance** state. • The call processing request towards MiCollab Unified Messaging has timed out. • All voice mail ports are in use.
<RAC>[1] IS UNPLUGGED	A memo request failed because the SIP peer voice mail is in the **Out-of-Service** or **Out-of-Service-Maintenance** state.
WAITING FOR *<RAC>*[1]	Follows the **<RAC> IS UNPLUGGED** message and indicates that memo request is camped-on and waiting for resources.
<RAC>[1] IS UNAVAILABLE	The memo request cannot camp-on.
DESTINATION IS NOT RESPONDING	The CP or SIP timer has timed out for a memo recording request.

1.   *<RAC> is* the Username for the RAC application as programmed in DB Programming.

TROUBLESHOOTING MICOLLAB UNIFIED MESSAGING RECORD-A-CALL

Table 158 provides troubleshooting information for MiCollab Unified Messaging RAC.

Table 158:   MiCollab Unified Messaging Record-A-Call Troubleshooting Issues

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Record-a-Call terminates automatically as soon as it is connected. The display message shows: • REQUESTING RECORD-A-CALL • RECORD-A-CALL IN PROGRESS • RECORD-A-CALL TERMINATED	The user requested an invalid mailbox when requesting RAC.	Verify that you have provisioned the user with a valid mailbox number in MiVoice Office 250 DB Programming.
	The user requested a valid mailbox but the corresponding mailbox does not exist on the MiCollab Unified Messaging system.	Log on to the MiCollab Unified Messaging administrator interface and provision the corresponding mailbox in the system.
	The MiCollab Unified Messaging mailbox is full.	Log on to the MiCollab Unified Messaging administrator interface and delete unnecessary messages on the MiCollab Unified Messaging mailbox to provide adequate storage for new messages.
	The MiCollab Unified Messaging mailbox does not have the RAC FCOS enabled.	Log on to the MiCollab Unified Messaging administrator interface and enable the RAC FCOS for the corresponding MiCollab Unified Messaging mailbox.
The RAC request does not succeed. The display message shows **RECORD-A-CALL IS REJECTED**.	There are not enough voice mail ports available on MiCollab Unified Messaging or all the voice mail ports are busy.	Log on to the MiCollab Unified Messaging administrator interface and verify the number of voice mail lines/ports assigned to Voice Mail Line Group. If necessary, configure the number of voice mail lines to a number that optimally meets call volume requirements
	All voice mail ports on MiCollab Unified Messaging are in use.	
	There are not enough IP resources available on the MiVoice Office 250.	Access DB Programming and verify that you configured IP resource reservations correctly. Make sure you have reserved the required number of IP resources for voice mail calls.
	All MiVoice Office 250 IP resources are in use.	
	There are not enough Conference resources available on the MiVoice Office 250.	When this error occurs, check the number of active conferences and RAC sessions. RAC sessions are limited by the number of conference resources available in the system.
	All MiVoice Office 250 conference resources are in use.	
	MiCollab Unified Messaging is not reachable and the SIP request towards MiCollab Unified Messaging timed out.	Verify the following: • The MiVoice Office 250 can connect to MiCollab Unified Messaging. • You have configured the correct IP address/hostname/port number for the SIP Voice Mail in DB Programming.

Table 158:   MiCollab Unified Messaging Record-A-Call Troubleshooting Issues (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
The RAC request does not succeed. The display message shows **CALL CANNOT BE COMPLETED**.	There are not enough voice mail ports configured on the MiVoice Office 250 to complete voice mail calls to MiCollab Unified Messaging.	Access DB Programming and configure sufficient ports for the corresponding SIP Voice Mail by specifying the value in the **Maximum Number of Ports** field. For example, if this value is set to **2** and there are two RAC or voice mail sessions already in progress, then new RAC requests will result in this error.
	The SIP Voice Mail for MiCollab Unified Messaging is in one of the following states: • Out-Of-Service • Out-Of-Service-Maintenance	Access DB Programming and change the SIP Voice Mail operating state to **In-Service**. The **Out-of-Service** operating state is reached if the **Keep-Alive** option is enabled for the SIP peer and the corresponding SIP peer (MiCollab Unified Messaging) does not respond to Keep-Alive requests. MiCollab Unified Messaging may not respond to keep-alive requests due to one of the following: • Network error. • SIP Peer IP Address/hostname/port number is not configured correctly.
	The RAC feature timed out while waiting for a response from MiCollab Unified Messaging.	Verify the following: • The MiVoice Office 250 can connect to MiCollab Unified Messaging. • You have configured the correct IP address/hostname/port number for the SIP Voice Mail in DB Programming.

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DIAGNOSTICS

The following tools provide diagnostics information for MiCollab Unified Messaging RAC:

- **Message Print**: When the user initiates a RAC request, MiVoice Office 250 Call Processing sends a SIP INVITE message towards MiCollab Unified Messaging. The diversion header in the SIP INVITE contains a reason parameter of RAC. The format for the SIP INVITE is as follows:

```
Diversion: <sip:extension@IP address>;Reason=RAC
```

You can view the SIP INVITE message using Message Print to verify that the reason parameter is RAC.

NOTE	To provide all of the SIP information to Message Print, the **SIP Message Output Format** option in DB Programming, must be set to **Full**. This OLM field is located under System\Devices and Feature Codes\SIP Peers\SIP Voice Mails\P9xxx**Configuration**. Do *not* use OLM mode unless you are instructed to do so by Mitel technical support personnel.

- **SIP Log Files**: The current SIP log file contains information about any SIP Message flows involved in SIP Record-a-Call application. You can access SIP Log files from the Logging tab on the MiVoice Office 250 Administrator Web Page.

NOTE	To provide SIP messages in the SIP log file, the **SIP Log Level** option in DB Programming, must be set to **Information** or **Debug**. This OLM field is located under System\Devices and Feature Codes\SIP Peers**General Configuration**. Do *not* use OLM mode unless you are instructed to do so by Mitel technical support personnel.

- **Information Dumps**: You can use the System Monitor – **Dump Extension** functionality to dump the information for the SIP RAC application extension. In the example below, the RAC application extension is 25154 and the pilot number extension is 25150. The RAC pilot number must be the same extension as the SIP Voice Mail application (see **page 900**).

```
-15:858- >
DUMP OF EXTENSION '25154'
-15:858- > --- sip_voice_mail_interface_entity_t:
-15:858- > --- sip_interface_entity_t:
-15:858- > --- entity_t:
-15:858- >       id:      ID(15:'25154')
-15:858- >       app_device_type:      1100
-15:858- > --- device_entity_t:
-15:858- >       username:      RAC
-15:858- >       pc_description:      Record-a-Call
-15:858- > --- devices_queued_onto_me_list_t:
-15:858- >       is empty
-15:858- > --- devices_queued_onto_me_list_t end
-15:858- >       ACD hunt groups logged into []: None

-15:858- >       ACD hunt groups not logged into []: None

-15:858- > --- physical_device_entity_t:
-15:858- >       language:      PRIMARY
-15:858- >       get_actual_extension():      'P9001'
-15:858- > --- sip_interface_entity_t end
-15:858- >       transfer_recall_destination:      '25154'
-15:858- >       attendant:      ''
-15:858- >       audio_for_calls_ringing_this_device:      2
-15:858- >       audio_for_calls_ringing_this_device_board_number:      8
-15:858- >       audio_for_calls_camped_onto_this_device:      3
-15:858- >       audio_for_calls_camped_onto_this_device_board_number:      8
-15:858- >       audio_for_calls_holding_for_this_device:      3
-15:858- >       audio_for_calls_holding_for_this_device_board_number:      8
-15:858- >       transfer_CID_propagation:      TRUE
-15:858- >       pilot_number:      '25150'
-15:858- >       prompt_for_mailbox_on_xfr:      FALSE
-15:858- >       validate_mailbox_on_xfr:      TRUE
-15:858- >       calling_party_number:      ''
-15:858- >       calling_party_name:
-15:858- >       day_cos_bit_list:
-15:858- >           Unrestricted
-15:858- >       night_cos_bit_list:
-15:858- >           Unrestricted
-15:858- >       day_user_group:      1
-15:858- >       night_user_group:      1
-15:858- > --- sip_voice_mail_interface_entity_t end

*** Command Completed ***
```

SCHEDULED TIME-BASED APPLICATION ROUTING (STAR)

This section contains the following information:

- **Feature Description** below

- Programming STAR Schedules on **page 913**

- Programming the Default STAR Application on **page 914**

FEATURE DESCRIPTION

STAR allows you to have applications with alternate greetings or different programming set up for holidays, weekends, and other scheduled events. Scheduled Time-Based Application Routing (STAR) allows you to have applications with alternate greetings or different programming set up for holidays, weekends, and other scheduled events.

A STAR application is basically a "routing table" for voice processor applications. When a direct ring-in call (from a trunk group or call routing table) rings in to a STAR application, it sends the call to another voice processor application, according to its programmed schedule. The caller is not aware of this transfer, but hears the programmed day or night greeting for the destination application. (The STAR application itself does not play a greeting.)

STAR can be used with any type of voice processor application except Auto Attendant Recall and Record-A-Call. You can even send calls from one STAR application to another, thereby "chaining" the applications together to increase the number of available schedules.

There can be as many STAR applications as desired, so long as the maximum limit for the number of voice processing applications is not exceeded.

STAR can be used with any type of Voice Processing application except Auto Attendant Recall and Record-A-Call. Calls can be sent from one STAR application to another, thereby "chaining" the applications to increase the number of available schedules. **Table 159** shows several STAR entry samples. A STAR application can contain up to 20 scheduling entries with the following fields:

- **Application:** This is the application (CRA, automated attendant, another STAR application, and so forth) that is used when the scheduling information applies to the incoming call.

- **Start/Stop Date:** If the schedule is going to be active on a single day or for a period of days, enable the Specific Date flag and enter Start and Stop Dates. (To have the schedule active on only one day, the Start and Stop Dates can be the same day.)

- **Days of the Week**: If the Specific Dates flag is disabled, the schedule can be used on specific days of the week.

- **Specific Times:** The schedule can be set to be active for a specific period of time on selected days, if you enable the Specific Times flag and enter a Start and Stop time.

- **Day/Night Mode:** The schedule can be set to be active in day or night mode, if the Specific Time flag is disabled.

Table 159:   Example STAR Entries

ENTRY	DESCRIPTION	DATE	DAY	TIME INTERVAL	DAY/ NIGHT	APPLICATI ON
1	Labor Day/ Summer Bank Holiday	Sep. 1/Aug. 28		10:00am - 2:00pm		2600
2	Labor Day/ Summer Bank Holiday	Sep. 1/Aug. 28			Both	2601
3	Memorial Day/ Spring Bank Holiday	May 30/May 29		8:00am - 5:00pm		2605
4	Memorial Day/ Spring Bank Holiday	May 30/May 29		5:00pm - 8:00am		2606
5	Christmas Day	Dec. 25			Both	2607
6	Christmas Holiday	Dec. 24 - Jan. 2			Both	2608
7	Weekends		Sat, Sun		Both	2612
8	Lunch Break		M - F	11:00am - 1:00pm		2613
9	Night Shift		M - F		Night	2614
10	Day Shift		M - F		Day	2615

- There are two entries for Labor Day and Memorial Day [Summer Bank Holiday and Spring Bank Holiday in Europe], and Christmas Day and Christmas Holiday overlap. The first entry in the table is checked first, and so forth, until a match is found. So it is important that you program the applications in the correct order. For example:

- The Labor Day [Summer Bank Holiday] entry, with a specific time, comes before the Day/ Night entry so that a different message is played from 10:00 AM– 2:00 PM only.

- The early time intervals come before the later intervals on Memorial Day [Spring Bank Holiday].

- Dec 25 comes before the range of dates Dec 24–Jan 4, so that the special holiday message is played on that day only.

When a match is found, the application corresponding to the matched entry is invoked. If no match is found, the application programmed in the *Default Application* field is invoked.

NOTE	When programming the days and times, be careful not to overlap. For example, if you place a date entry for Dec 25 after a day of the week entry for Monday, and Christmas is on a Monday, the "Monday" application is selected instead of the "Dec 25" application.

PROGRAMMING STAR SCHEDULES

Each schedule can have a description. To enter the description, select the current Value and then enter a description for the schedule, up to 20 characters. To program the remaining information for each schedule, double-click the schedule to display the following fields:

- **Application:** This is the extension number of the application that will be invoked when the time-based information in the table matches that of the incoming call. To assign the application, use one of the following methods:

  Method A

  a. Select the current Value, and then enter the new value in the text box.

  b. Press **ENTER**. A screen appears displaying what is associated with the number entered.

  c. Click **OK**. The new number appears in the field.

  Method B

  a. Right-click the current Value and select **Change Application**. A window appears prompting for the type to include.

  b. Select the desired application, and then click **Next**. A list of applications appears. You can view them in a list by selecting the List button or view details by selecting the Details button.

  c. Select the desired application, and then click **Finish**. The new value appears in the field. To view programming options, double-click the application.

- **Specific Date:** If you want the schedule to be active on a single day or for a period of days, enable this flag.

  To enable this flag:

  a. Select the check box. The field changes to **Yes**. To disable the option, clear the check box.

  b. Click out of the field or press **ENTER** to save the change.

- **Start and Stop Date:** If the Specific Date field is set to **Yes**, use these fields to set the date(s) that this schedule will be active (such as a holiday).

  a. Enter the date you want the schedule to begin in the **Start Date** box and the date you want it to end in the **Stop Date** box. To have the schedule active on only one day, set the Start and Stop Dates to the same day.

  b. After entering information, click out of the field or press **ENTER** to save the changes.

- **Days of the Week:** If the Specific Date field is set to **No**, you can use these fields to determine which days of the week the schedule will be active.

  To enable a specific day:

  a. Select its check box to place a mark in it and change the setting to **Yes**. To disable the day, select the check box again to remove the mark and set it to **No**.

  b. Click out of the field or press **ENTER** to save the changes.

- **Specific Times:** If you want the schedule to be active for a specific period of time on the selected day(s), enable this option.

  To enable this option:

  a. Select the check box. The field changes to **Yes**. To disable the option, clear the check box.

  b. Click out of the field or press **ENTER** to save the change.

- **Start and Stop Time:** If the Specific Time field is set to **Yes**, use these fields to set time period that this schedule will be active (such as a after hours).

  a. Enter the time you want the schedule to begin in the **Start Time** box and the time you want it to end in the **Stop Time** box.

  b. After entering information, press **ENTER** or click another field to save the changes.

- **Day/Night Mode:** If the Specific Time field is set to **No**, you can use this field to determine whether the schedule will be active in day and/or night mode.

  To change this field:

  a. Select the option from the list.

  b. Click out of the field or press **ENTER** to save the change.

PROGRAMMING THE DEFAULT STAR APPLICATION

The default application is where calls go that do not match any of the time-based criteria in tables 1–20.

To assign the application, use one of the following methods:

Method A

1. Select the current Value, then enter the new value in the text box.

2. Press **ENTER**. A screen appears displaying what is associated with the number entered.

3. Click **OK**. The new number appears in the field.

Method B

1. Right-click the current Value and select **Change Application**. A window appears prompting for the type to include.

2. Select the desired application, then click **Next**. A list of applications appears. You can view them in a list by selecting **List** or view details by selecting **Details**.

3. Select the desired application, then click **Finish**. The new value appears in the field. To view programming options, double-click the application.

VOICE MAIL (APPLICATION)

This section contains the following information:

- **Feature Description** below
- **Programming Voice Mail Applications** below

FEATURE DESCRIPTION

The Voice Mail application handles all calls that are directed to voice mail by subscribers and non-subscribers, other than to the Message Notification/Retrieval application (see **page 890**). Callers hear the main company greeting, followed by a menu of available options. Phones can forward or transfer calls directly to their mailbox using this application extension number.

You can create up to 1,000 voice mail applications. With more than several hundred voice mail applications, the system will take longer to come up to the operational state. For instance, DB Programming may not be able to connect for 10–20 minutes until Call Processing and Voice Mail have acknowledged all of these applications. Also, system monitor dumps (performed though the Diagnostics Monitor app) may take a few minutes to complete. As a workaround, create these applications in offline mode, then restore the database.

For more information about voice processing applications, see "Voice Processor Applications" on **page 870**.

PROGRAMMING VOICE MAIL APPLICATIONS

This section describes the following programming information

- **Day and Night Greetings** below
- Attendants on **page 916**
- Music-On-Hold on **page 917**
- Time Slot Group on **page 917**
- Transfer Recall Destination on **page 918**
- Propagate Original Caller ID on Transfer on **page 919**
- Calling Party Name and Number on **page 919**

DAY AND NIGHT GREETINGS

Applies to UVM voice processors only. When the application receives a call, a custom audiotex recording plays. For more information about creating audiotex recordings, refer to the *MiVoice Office 250 Unified Voice Messaging Administrator Guide,* part number 580.8009. You can select any greeting number for the Day and Night Greetings. The Day message is played when the system is in day mode, and the night message is played during night mode. The greetings can be changed or you can use the default greetings.

When used on Automated Attendants and CRAs, the greeting can include pauses and can announce queue position and time to wait using special audiotex selections. The special characters that can be used for programming greetings include the following:

- **Short Pause**: Inserts a 1-second pause.

- **Intermediate Pause**: Inserts a 5-second pause.

- **Long Pause**: Inserts a 10-second pause.

- **Position in Queue**: *(Used for hunt group Call Routing Announcements only.)* Tells the caller how many calls are waiting ahead of his or her call.

- **Time To Wait**: *(Used for hunt group Call Routing Announcements only.)* Tells the caller how long he or she can expect to wait, based on the number of calls waiting and the Average Call Length programmed for the hunt group.

To add or change a greeting:

1. Double-click **Day Greeting** or **Night Greeting** to view the current list of greetings.

2. *To add a a greeting above an existing one*, select that greeting, right-click, and then select **Add To Day Greeting List** or **Add To Night Greeting List**.
   To add to the end of the list, right-click in any blank area of the screen and then select **Add to Greeting List** or **Add To Night Greeting List**. A window appears prompting for the type to include.

3. Select **Audiotex Recording**, and then click **Next**. The list of Audiotex Recordings with details appears. To view items in a list only, click **List**.

4. Select the recording(s) and/or special characters or pauses you want to use, and then select **Add Items**. The selections appear in the list. When done, click **Finish**.

To move a greeting to another location in the list:

1. Double-click **Day Greeting** or **Night Greeting** to view the current list of greetings.

2. Drag and drop the greeting to the new position. Or, select the greeting to move and press **CTRL** + the up/down arrow to move the greeting up or down in the list.

To delete a greeting:

1. Double-click **Day Greeting** or **Night Greeting** to view the current list of greetings.

2. Select the greeting(s).

3. Right-click and select **Remove Selected Items**. To select a series of recordings, hold down **SHIFT** while selecting the first and last recordings in the range. To select two or more recordings that are not consecutive, hold down **CTRL** while selecting the desired greetings.

ATTENDANTS

Applies to UVM voice processors only. A phone, hunt group, or other application can serve as the attendant phone for an application. This attendant will receive recalls when the Transfer Recall Destination does not answer or is unavailable.

To program the attendant for the application, use one of the following methods:

Method A

a. Select the current value, and then enter the new value in the box.

b. Press **ENTER**. A screen appears displaying what is associated with the number entered.

c. Click **OK**. The new number appears in the field.

Method B

a. Right-click **Attendant** and select **Change Attendant**. A window appears prompting for the type to include.

b. Use the drop-down list box to scroll to **Keyset**, and then click **Next**. A list of phones with details appears. To view the items in a list only, click **List**.

c. Select the desired phone, and then click **Finish**. The new attendant appears in the Attendant field. To view programming options, double-click the attendant.

MUSIC-ON-HOLD

Applies to UVM voice processors only. **You can program the following audio options that callers hear when waiting for system users:**

• **Audio for Calls Camped onto this Device**

• Audio for Calls Holding for this Device

• Audio for Calls Ringing this Device

For more information about these options, see "Device Audio for Calls Settings" on **page 414**.

To program Music-On-Hold for applications:

1. Select Voice Processor – Devices – **Applications**.

2. Double-click the application.

3. Select the audio type.

TIME SLOT GROUP

Applies to UVM voice processors only. You must program Time Slot Groups to specify the total number of voice channels available for each application for processing calls. See "Voice Processor System Settings" on **page 862**.

To select the Time Slot Group for the application:

1. Select Voice Processor – Devices – **Applications**.

2. Double-click the application.

3. Right-click **Time Slot Group**, and then select **Change Time Slot Group**. A window appears prompting for the type to include.

4. Select **Time Slot Group**, and then click **Next**. A list of time slot groups with details appears. To view items in a list only, click **List**.

5. Select the desired group, and then click **Finish**. The new time slot group appears in the field. To view programming options, double-click the time slot group.

TRANSFER RECALL DESTINATION

UVM systems only. This is the phone, hunt group, or application that receives any calls that recall after being transferred by the application. For the Auto Attendant, this is usually the Automated Attendant Recall application.

The system can be programmed to determine the music source a caller hears based on the device for which the caller is waiting. By default, the system determines the music source based on the trunk group on which the call resides.

The following fields are programmable for each voice processor application:

- **Audio for Calls Camped onto this Device**: Defines the audio that a caller who is camped-on to the device hears.

- **Audio for Calls Holding for this Device**: Defines the audio that a caller who is holding for the device hears.

- **Audio for Calls Ringing this Device**: Defines the audio that a caller who is ringing the device hears. The default is *Ringback*.

NOTE	If the trunk group audio field, including Music-On-Hold, is set to Use Next Device Audio Source, the system uses the programming for the next device as programmed for the Day/Night trunk group destination. If the field is set to any other option, the system uses the trunk group audio source, overriding phone programming. The Audio for Calls Ringing this Device field only works when the call goes through a trunk group and also when used in conjunction with the "use other device source" field. IC calls do not apply to the use of this field when this field is set to a music source. For a hunt group whose primary purpose is to support IC callers (for example, an internal help desk), you should set all of the "Audio for Calls..." fields to something other than a music source, such as Ringback.

When a caller on a node waits for a device on a node, the caller hears the music source programmed for the node trunk group on the destination node.

To select the Transfer Recall Destination for the application, use one of the following methods:

Method A

1. Select Voice Processor – Devices – Applications – *<application>* – **Transfer Recall Destination**.

2. Select the current value, and then enter the new value in the text box.

3. Press **ENTER**. A screen appears displaying what is associated with the number entered.

4. Click **OK**. The new number appears in the field.

Method B

1. Select Voice Processor – Devices – Applications – <***application***>.

2. Right-click **Transfer Recall Destination** and select **Change Transfer Recall Destination**. A window appears prompting for the type to include.

3. Select **Auto Attendant Recall** or your desired destination, then click **Next**. A list of Auto Attendant recall applications appears. You can view them in a list by selecting the List button or view details by selecting the Details button.

4. Select the desired application, then click **Finish**. The new destination appears in the Transfer Recall Destination field. To view programming options, double-click **Transfer Recall Destination**.

PROPAGATE ORIGINAL CALLER ID ON TRANSFER

Applies to MNR only. When this option is enabled, if the phone is on a call with an outside trunk that had caller ID information, the application propagates the caller ID when a voice mail application transfers a call to a phone and the phone performs a transfer back to the PSTN. To propagate the caller ID to the PSTN, the eventual trunk must be ISDN and the "Propagate Original Caller ID" flag in its CO trunk group must be set to **Yes**. The default of this flag is set to Yes (which means propagate caller ID on transfer PSTN calls). The application must be able to perform transfers. For details, "Caller ID Propagation" on **page 176**.

To propagate caller ID when a Voice Mail application transfers a PSTN call:

1. Select Voice Processor – Devices – Applications – <***application***>.

2. Set the **Propagate Original Caller ID on Transfer** flag to **Yes**.

3. Press **ENTER** or click another field to save the change.

CALLING PARTY NAME AND NUMBER

You can enable the Calling Party Name and Calling Party Number features for the MNR application. For feature descriptions, see "Calling Party Name" on **page 411** and "Calling Party Number" on **page 411**.

1. Select Voice Processor – Devices – Applications – <***application***>.

2. Select **Calling Party Name** or **Calling Party Number**.

3. In the **Value** column, select the current value, and then type the name in the box.

4. Click out of the field or press **ENTER** to save the change.

EXTENSION IDS

Applies to UVM voice processors only. Extension IDs provide the Auto Attendant application with a means for transferring calls to extensions and off-node devices that do not have mailboxes. An extension ID allows the owner to record a name for the directory and establish a passcode. Extension IDs can be created for phones, hunt groups, modems, and applications. To create a mailbox for an extension that currently has an extension ID, first delete the extension ID, then create the mailbox. Extension IDs cannot be created for off-node device wildcard extensions.

Extension IDs are used when transferring calls through the Automated Attendant or using the Extension Directory. The extension ID allows callers to be transferred to phones and applications that do not have mailboxes. It also allows the phone or application to have a recorded name in the directory.

If an Extension ID has been created in database programming for a phone extension number, either the principal owner of the extension or the Voice Mail System Administrator must set up (initialize) the ID with a new passcode and record a name for use in the Extension Directory. At default, the Extension ID passcode is the same as the extension number.

If the Extension ID has not been "initialized," calls can still be transferred to the associated extension number. However, they cannot be accessed from the directory. If the Extension ID user name has not been recorded, it cannot be heard when callers access the Extension Directory. The name must be recorded to fully initialize the Extension ID.

NOTE	To provide system security, all extension IDs should have a passcode. To make the passcodes difficult to guess, they should not match the phone extension number or consist of one digit repeated several times. The default passcode should be changed the first time the user logs in.

After a passcode has been set up and the name recorded, the extension owner may access Extension ID Options which allows the associated directory name and passcode to be modified.

If an Extension ID has not been created for a phone, callers using the automated attendant cannot be transferred to that destination. Instead, these callers receive a system recording notifying them of an invalid entry and are routed back to the automated attendant's main menu.

Individual extension IDs can be programmed as "Private" in database programming. A Private number is included in the directory and can be dialed if the caller knows the number, but the extension number information is not available to the caller; only the name is played.

Extension IDs can also be programmed as "Unlisted." That means that the number is not included in the directory but can be dialed if the caller knows the extension number.

Calls transferred from the Automated Attendant or a CRA application to phones with extension IDs can be screened, announced, or unannounced. Programming flags determine the methods used for transferring calls to phones with extensions IDs. In a network setting, the external voice processing system cannot create extension IDs for off-node phones included in wildcard ranges. Each associated phone must have an off-node device entry.

To create an Extension ID:

1. Right-click in any area of the screen and select **Create Extension ID**. A window appears prompting for the device type to include.

2. Select the device type, and then click **Next**. The list of devices appears. You can view them in a list by selecting **List** or view details by selecting **Detail**.

3. Select the devices for which you want to create IDs.

4. Select **Add Items**. The selections appear in the mailbox list. (You can use the SHIFT or CTRL keys to select more than one item.)

5. Click **Finish** to exit.

To delete an Extension ID:

1. Select the IDs.

2. Right-click, and then select **Delete**.

Extension ID programming includes the following fields:

*   Allow Transfer Method Programming

*   Auto Attendant Transfer Prompt

*   Unlisted Number

*   Private Extension

*   Passcode

*   Transfer Method

To program an Extension ID: The description and username fields cannot be changed. They are programmed as part of device programming.

*   *Allow Transfer Method Programming:* Determines whether the extension ID user (or the voice mail System Administrator) will be allowed to change the Transfer Method, using the voice mail Personal Options prompts.

    a. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

    b. Click out of the field or press **ENTER** to save the change.

*   *Auto Attendant Transfer Prompt:* Determines whether the transfer prompt ("Please hold while your call is being transferred to...") plays after a caller has entered the extension number of the phone associated with this extension ID. This applies to calls transferred by Automated Attendant and Call Routing Announcement applications, including transfers to the operator's mailbox or extension ID.

*   To disable the prompt:

    a. In the **Value** column, clear the check box. The field changes to **No**. To enable the option, select the check box.

    b. Click out of the field or press **ENTER** to save the change.

- *Unlisted Number/Private Extension:* Unlisted numbers are not included in the directory, but can be dialed if the caller knows the extension number. Private numbers can be dialed, but only the name is played in the directory.

  a. In the **Value** column, select the check boxes to enable the options. The fields change to **Yes**. To disable the options, clear the check boxes.

  b. Click out of the field or press **ENTER** to save the change.

- *Passcode:* To program a passcode for the Extension ID:

  a. Right-click the Passcode field and select **Edit Passcode**. The Edit Passcode dialog box below appears.

  b. In the **New Passcode** box, type the new passcode (up to 12 digits, using digits 0–9). Typed characters appear as asterisks (***).

  c. Retype the passcode in the Confirm Passcode box.

  d. Click **OK** to exit and save the passcode. If the entered passcodes match, you will return to the Passcode field. If they do not match, you must re-enter the new passcode and verify it again. If you make a mistake while entering the passcode or want to leave it unchanged, select **Cancel**.

  > **NOTE** To provide system security, *all* extension IDs should have a passcode. To make the passcodes difficult to guess, they should ***not*** match the mailbox number or consist of one digit repeated several times.

- *Transfer Method:* Determines how transfers will be made to this extension ID. The options are: Announce Only, Screened, and Unannounced. The default is set to Unannounced. For more information, see "Call Screening" on **page 279**.

  a. Select the option from the list.

  b. Click out of the field or press **ENTER** to save the change.

GROUP LISTS

UVM systems only. Group lists are lists of mailboxes that can be used by any subscriber for sending messages to several mailboxes simultaneously. The information that is programmed for group lists include the following:

- Group list description

- Group list number

- Mailboxes included in the list

There can be up to 1000 group lists in the system and up to 1500 entries per group list. Group Lists are not included in the Automated Attendant or Voice Mail directories.

CREATING A GROUP LIST

To create a Group List:

1. Select Voice Processor – Devices – **Group Lists**.

2. Right-click anywhere in the right pane. The Create Group List Extension dialog box appears.

3. Enter the Group List number, and then click **OK**.

4. Type a description of up to 20 characters in the Description text box.

5. Type a name of up to 10 characters in the Username text box. This is the name that will appear on phone displays. Do *not* use slash (/), backslash (\), vertical slash (|) or tilde (~) characters in usernames. Do *not* use control characters in descriptions or usernames.

CHANGING A GROUP LIST EXTENSION NUMBER

To change a Group List Extension number:

1. Select Voice Processor – Devices – **Group Lists**.

2. Type the desired number in the text box *or* use the arrow to scroll to an available number. If you attempt to enter an invalid number, a number that conflicts with an existing extension, or a number that is already assigned, you will see a warning window that allows you to fix the error. Type a new number, and then click **OK** to continue.

3. Press **ENTER** or click another field to save the change.

ADDING MAILBOXES TO GROUP LISTS

To determine the mailboxes that will be included in the Group List, double-click the Group List entry. The current list of mailboxes appears in the right screen.

To add a mailbox to the Group List:

1. Select Voice Processor – Devices – **Group Lists**.

2. Right-click in any area of the screen and select **Add To List**. A window appears prompting for the type to include.

3. Select **Mailbox** and/or **Off-Node Mailbox**, and then click **Next**. The list of mailboxes appears. You can view them in a list by selecting the **List** button or view details by selecting the **Details** button.

4. Select the mailboxes you want to add, then select **Add Items**. The selections appear in the list.

5. Click **Finish** to exit.

REMOVING MAILBOXES FROM GROUP LISTS

To remove a mailbox from the list:

1. Select Voice Processor – Devices – **Group Lists**.

2. Select the mailboxes that you want to remove.

3. Right-click and select **Remove Selected Items**. The information is automatically removed from the list. To select a series of mailboxes, hold down **SHIFT** while selecting the first and last mailboxes in the range. To select two or more that are not next in consecutive order, hold down **CTRL** while selecting the desired mailboxes.

VIEWING GROUP LIST MEMBERS

Instead of the members of a group list being displayed directly in the individual group list, the members reside in a separate folder called Members, as shown in the following two illustrations.
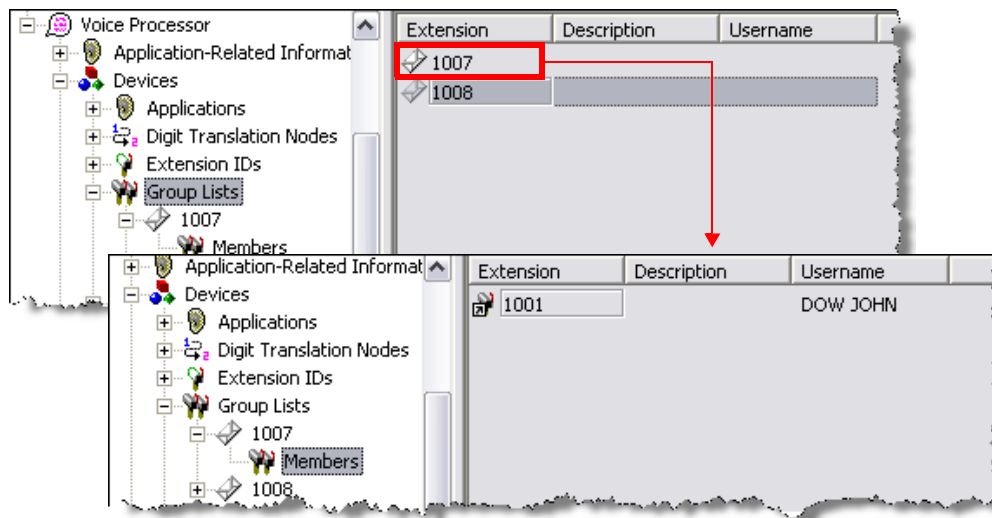
AUDIOTEX RECORDINGS

Applies to UVM voice processors only. Audiotex recordings are custom recordings used by voice processing applications. Audiotex recordings are recorded using a phone and the voicemail administrator mailbox. For more information about creating and changing Audiotex recordings, refer to the *MiVoice Office 250 Unified Voice Messaging Administrator Guide,* part number 580.8009.

The recording length is shown in seconds for reference only. You cannot change the recording time.

For VPUs, this programming area allows you to assign a description to each Audiotex recording. It also allows you to view the length of each recording.

NOTE	You can view recording information (recording number, description, and recording length) for audiotex recordings that have been re-recorded.

To change the description of a recording:

1. Select – Voice Processor – Application-Related Information – **Audiotex Recordings**.

2. Click once on the current description.

3. Type the new description in the box, using up to 20 characters. If desired, you can right-click in the box and cut, paste, copy, or delete the description.

4. Click out of the field or press **ENTER** to save the change.

5. When the current description is selected in the text box, right-click and select the desired operation.

DIRECTORIES

UVM systems only. There are two types of Voice Processing directories that can be enabled (or disabled) in the system: Mailbox and Extension.

The Voice Mailbox Directory: A list of mailbox subscribers, their recorded names, and mailbox numbers. For programming instructions, see **page 929**.

The Automated Attendant Directory: Provided to all Auto Attendant callers, this is a list of all mailbox subscribers and extension ID owners and their recorded names. For programming instructions, see **page 876**.

The directories can be selected in the following ways:

- **From the Voice Mail or Automated Attendant main menu**: Press the **#** button.

- **Using a CRA application**: Press the Digit Translation digit designated for directory access. For a detailed explanation of Digit Translation, see **page 884**.

- **As a Voice Mail Subscriber**: Subscribers can access the directory by pressing **#** whenever they are prompted to enter a mailbox number.

If a directory is disabled or empty because no names are recorded for any of the system's mailboxes and extension IDs, callers do not hear the prompt that allows access to the directory. However, if the caller does press **#** or a Directory menu button, the user is instructed that the selection is invalid and returned to the initial instructions.

Recorded names are added to the directories when the owner of the mailbox or extension ID has initialized the name. If the mailbox or ID is not initialized, the directory includes only the mailbox number or extension ID number. (Group Lists are not included in either type of directory.)

The caller uses the keypad buttons to enter the name. The application then plays the closest matching directory name that corresponds to the digits entered by a caller. Once the name has been played, the system returns a menu of options, including the following:

- Listening to the previous or next name in the directory

- Listening to additional information (not available if the mailbox or extension ID number has been classified in the database as a "Private" number)

- Spelling a new name

- Toggling from a last to first name search mode

- Accepting the name

LOCATING A NAME

This section details the following methods to locate a name:

- **Entering a Name** below

- Changing the First/Last Name Search on **page 927**

- Listening to the Next/Previous Name on **page 928**

For end-user instructions, refer to the *Unified Voice Messaging, and Embedded Voice Mail Card User Guide*, part number 835.3205.

ENTERING A NAME

There are two methods used for entering a name: Quick Spell and Exact Spell. Outside callers and phone users with non-display phones and single-line sets use Quick Spell. Display phones use the Exact Spell method.

- **Quick Spell**: Callers press a single digit (or button) from their digital telephone dial pad for each letter or character entered. For example, keypad button **2** shows ABC, button **3** shows DEF, and so forth. To enter JONES, you would press **5 6 6 3 7**. Some characters are not shown on the buttons: for "Q" press **7**, for "Z" press **9**, for punctuation marks press the **1** button.

- **Exact Spell**: Callers press the keypad buttons to enter the name. The number of times a button is pressed determines which character is entered, as shown on the following table. When adjoining characters are located under the same button, press **FWD** once to advance to the next character. For example, 5666 **FWD** 66337777 would enter JONES.

NUMBER OF TIMES BUTTON IS PRESSED

BUTTON	1	2	3	4	5
1	-	&	(	)	1
2	A	B	C	'	2
3	D	E	F	!	3
4	G	H	I	*	4
5	J	K	L	#	5
6	M	N	O	Ñ	6
7	P	Q	R	S	7
8	T	U	V	?	8
9	W	X	Y	Z	9
0	@	:	.	,	0

After the digits have been entered, the caller presses **#** to begin the search. The Automated Attendant application plays the name that most closely matches the digit(s) that were entered. If **#** is pressed without entering any digits, the caller hears the first name in the directory.

If a user presses **0** at any time while spelling a name, the system plays a prompt instructing the caller how to enter a name. (Display phone users receive Exact Spell instructions; all other users hear Quick Spell instructions.)

CHANGING THE FIRST/LAST NAME SEARCH

Each directory can be programmed to be organized by last name or by first name. Callers accessing a directory receive system voice prompts that ask them to enter the first or last name of the person for whom they wish to leave a message.

If a phone with menu buttons is used to access a directory, the caller may switch back and forth from looking up a last name, to a first name simply by pressing the corresponding menu button. Callers using Standard Display, Associate Display, or 8520 phones or single-line sets can press **5** to toggle between last and first name.

LISTENING TO THE NEXT/PREVIOUS NAME

After a name has been played, the caller can press **1** to listen to the previous name in the directory. To listen to the next name, the caller can press **3**.

The directory lists are circular. That is, when the end of the list is reached, the next name played is the first name in the directory. Or, if the caller scrolls to the beginning of the list, the "previous" name played is the last name in the directory.

ACCEPTING A NAME

When the caller is using the Voice Mail feature and accepts a name, the caller is transferred to the corresponding mailbox. Then the caller hears either the subscriber's recorded greeting, recorded directory name, or, if no recording has been made, the "mailbox number XXX is not available" prompt. The caller can then record a message.

If the caller is using the Automated Attendant feature and accepts the name, the caller is transferred to the selected destination (phone or mailbox) if it is available. If a phone extension is dialed and the destination is not available, the caller is sent to the associated mailbox, if one exists.

REQUESTING ADDITIONAL INFORMATION

After a name is selected from the directory, the caller has the option of requesting additional information. When requested, the mailbox number, extension number, and name are played (if they have not been disallowed).

Individual mailboxes and extension IDs can be programmed as "Private" in database programming. A Private number is included in the directory and can be dialed by a caller who knows the number, but the associated mailbox and extension number information are not available to the caller; only the name is played.

Mailboxes and extension IDs can also be programmed as "Unlisted." That means that the number is not included in the directory but can be dialed if the caller knows the extension number.

VOICE MAIL DIRECTORY

Applies to UVM voice processors only. The following options enable or disable the voice mail directory and determine the directory sort order.

ENABLING OR DISABLING THE VOICE MAIL DIRECTORY

If you enable the Voice Mail Directory, the system allows anyone routed or answered by voice mail or any mailbox users while in their mailbox to be presented with the directory option. If the flag is disabled, the voice mail will not prompt the caller for the directory option while in the voice mail areas (non AA or CRA).

NOTE	If the voice mail mailbox directory is disabled, callers using the voice mail system do *not* receive a system prompt giving the option to search the directory for the person to whom they want to speak. If the caller presses the dialpad button normally associated with access to the mailbox directory, the caller is informed that the selection is invalid.

To enable the Voice Mail Directory:

1. Select – Voice Processor – Application-Related Information – **Voice Mail Information**.

2. Select **Enable Voice Mail Directory**.

3. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

4. Click out of the option or press **ENTER** to save the change.

CHANGING THE VOICE MAIL DIRECTORY SORT ORDER

This option is for voice mail calls only. You can change the directory sort order, or whether the directory searches by subscribers' first or last names. The Directory Search Order also determines which system voice prompt plays (first or last name) when directing callers to spell a name. To change the Auto Attendant Sort Order, see "Auto Attendant Directory Sort Order" on **page 876**.

To change the Directory Sort Order:

1. Select – Voice Processor – Application-Related Information – **Voice Mail Information**.

2. Select **Directory Sort Order**.

3. From the **Value** column, select the option from the list.

4. Click out of the option or press **ENTER** to save the change.

VOICE PROCESSOR TIMERS AND LIMITS

Programmable timers and limits control various Voice Processor functions. The timers and their default values, programmable ranges, and purposes are described in the following paragraphs. The default values have been selected to ensure correct operation under most circumstances. Occasionally, one or more of the timers may need to be adjusted.

The Number of Voice Channels for UVM is programmed in the Voice Processor Timers and Limits folder.

The Timers and Limits folder contains the following programming areas:

- Timers and Limits on **page 859**

- DTMF Detection Information on **page 934**

- DTMF Generation Information on **page 937**

- Number of Voice Channels on **page 938**

Figure 3: Voice Processor Timers and Limits

PROGRAMMING UVM TIMERS AND LIMITS

To view the list of current values for the Voice Processor, select **Timers and Limits**.

To program a timer or limit:

1. Select Voice Processor – **Timers and Limits**.

2. Double-click **Timers and Limits**.

3. In the **Value** column, click the current value, and then type or select the new value.

4. Click out of the field or press **ENTER** to save the change.

Table 160 shows the system-wide timers and limits that can be programmed for the Voice Processor.

Table 160: Voice Processor Timers and Limits

TIMER	DEFAULT	RANGE	DEFINES
Busy Tone Cycle Detect	2 cycles	1–60 cycles	The minimum number of cycles of tone the system needs to recognize busy, Do-Not-Disturb, or reorder tones. When the voice processor does not recognize one of these tones, it will assume the call is answered. This timer is **not** supported in Unified Voice Messaging (UVM).
Call-in-Progress Dial Tone	2 sec	0–5 sec	The minimum duration of continuous dial tone that the voice processor can recognize during an active call (for example, when a caller hangs up while connected to the voice mail application). To disable dial tone detection on active calls, set this timer to 0. This timer is **not** supported in UVM.
Call Progress Delay	6000 msec	1–60000 msec. (0.001– 60 sec.)	The maximum length of time the voice processor will wait before checking call status when placing a call. It must be set long enough to prevent the voice processor from detecting dial tone on the outgoing call.
Call Progress Detection	4000 hundredths	1-6000 hundredths (0.01–60 sec.)	The maximum length of time the voice processor will wait for a call to be answered before aborting the attempted call. If it detects anything other than silence during this time, the system will consider the call answered. This timer is **not** supported in UVM.
Deleted Message Hold Duration	1 hour	0–24 hours	The amount of time that the voice processor will keep each deleted message. The duration is specified in hours and is based on the time the user deletes the message. For example, if the Deleted Message Hold Duration is set to 6 hours, and a user deletes a message at 9:22 AM, the voice processor will keep the message until 3:22 PM. A value of 0 means that the voice processor will not hold messages. (That is, as soon as the user deletes the message, it is permanently deleted and there is no way to restore it).
Dialed Pause Duration	300 hundredths	1–500 hundredths (0.01–5 sec.)	The duration of the pauses dialed by the voice processor as part of feature codes, numbers, and outgoing dialing strings. This timer is **not** supported in UVM.

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Table 160:   Voice Processor Timers and Limits (continued)

TIMER	DEFAULT	RANGE	DEFINES
Maximum Deleted Message Space	0 MB	0–20 MB	The maximum amount of space that the voice processor allocates to store deleted messages. The deleted message storage size is specified in megabytes (MB) of the hard disk. When a user deletes a message, and the deleted message queue size exceeds this number, the voice processor will purge the oldest message to make room for the new one. In this case, the voice processor will purge as many messages from storage as necessary to make room for the new one. However, if the hard disk becomes full, the voice processor will immediately purge all stored messages.
			Note: When Voice Processing is shut down, all deleted messages are permanently deleted and cannot be restored.
Maximum Greeting Length	1 min	1–15 min	The maximum time allowed for all mailbox greetings and custom audiotex recordings.
Maximum Number of Deleted Messages	0 messages	0–9999 messages	The maximum number of deleted messages that the voice processor will store. If this number is exceeded, and a user deletes a message, the voice processor will remove the oldest message in storage to make room for the new one. (Entering a value of 0 means that the number is unlimited.) However, this limit is bound to the availability of hard disk space as specified by the Maximum Deleted Message Space above. That is, the Maximum Deleted Message Space limitation will take precedence over this one.
Maximum Outgoing Calls	2 calls	1–number of available ports (max 32)	The number of outgoing message notification and fax delivery calls that can be made at one time by the Message Notification/Retrieval application. Note that the number of calls that can be made are also limited by the availability of intercom voice channels and outgoing trunks.
Minimum Call Progress Silence Duration	15 hundredths	1–21 hundredths (0.01–0.21 sec)	The minimum duration of a period of silence necessary for the voice processor to detect silence. That is, if a period of sound is broken by silence, and the silence is shorter than this timer, the voice processor will ignore the silence. This timer applies to outgoing remote notification calls only.
			This timer is **not** supported in UVM.
Minimum Call Progress Signal Duration	18 hundredths	1–18 hundredths (0.01–0.18 sec)	The minimum duration of a period of sound necessary for the voice processor to detect sound. That is, if a period of silence is broken by a burst of sound on the line, and that sound is shorter than this timer, the voice processor will ignore the sound. This timer applies to outgoing remote notification calls only.
			This timer is **not** supported in UVM.

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Table 160:   Voice Processor Timers and Limits (continued)

TIMER	DEFAULT	RANGE	DEFINES
Notification No-Answer Detection	4 rings per prompt iteration	1–25 rings 1–5 prompt iterations (automatic)	The number of rings the voice processor waits to deliver a remote message before trying to access an Alternate Notification mail box. Automatically sets the number of times, or iterations, the Primary and Alternate Notification Cascades are tried. Sets the iterations equal to the ring Value up to 4. Ring Value settings 5–25 do not exceed 5 iterations.
Off-hook Delay	0 seconds	0–5 seconds	Length of time the voice processor should wait after coming off hook before playing the first prompt. However, the recommended minimum setting is 1 second to allow a pause before the prompts are played for outside callers, thereby assuring that no text is cut off. It is set to 0 to allow internal callers to hear the prompts immediately.
Outgoing DTMF Digit Duration	60 msec	30–250 msec (0.030–0.250 sec)	Length of the DTMF tones (and inter-digit pauses) that are sent during remote notification to a pager or personal number. If this timer is changed, the PC must be reset. This timer is **not** supported in UVM.
Pause Voice Mail	30 sec	1–240 sec	The maximum amount of time that the voice processor will pause during the playback of a message or recording.
Record-A-Call Max Message Length	30 min	1–600 min	The maximum time allowed for Record-A-Call input to a mailbox. (This overrides the mailbox's maximum message length setting. However, if a Record-A-Call message exceeds the maximum message length, the mailbox will be considered full until that Record-A-Call message is deleted.)
Recording Beep Length	700 ms	100–2000 ms	The length of the tone played to indicate that recording has started.
Recording Beep Level	-20 dBm0	-63 to 0 dBm0	The level dBm0 of the tone played to indicate that recording has started.
Recording Silence Detection	5 sec	0–30 sec	Length of silence required to terminate a recording. If silence in recording exceeds this timer, the recording will automatically end. If lengthy (silent) pauses are included in any recordings, this timer should be adjusted to allow them.
Recording Silence Threshold	-38 dBm0	-40 to -20 dBm0;	Audio recorded below this level is considered silence by the system for the purposes of the recording silence detection timer.
Relay Forward/Rewind Increment	5 sec	1–60 sec	The number of seconds a message or recording will be advanced or backed up when a user skips ahead or backward during a replay.

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Table 160:   Voice Processor Timers and Limits (continued)

TIMER	DEFAULT	RANGE	DEFINES
Shortest Message Allowed	3 sec	1–5 sec	Length of the shortest message that will be accepted by the voice processor. (Messages terminated with # are always allowed, regardless of length.)
Transfer No-Answer Detection	4 rings	1–25 rings	The number of rings during which the voice processor will wait for a transferred call to be answered. Establishes the duration required for call-screening.

DTMF DETECTION INFORMATION

Applies to UVM only. In most installations, you should not need to change the DTMF detection information. However, if the voice processor is losing digits or if DTMF detection is causing talk-off, you may need to adjust the DTMF filter parameters. If digits are not being detected properly, there are three possible reasons:

- **The "twist" is not within the allowed range**: The Twist parameters specify the maximum acceptable difference between the high frequency energy and the low frequency energy. One frequency can be much stronger than the other, or the frequencies can be off the nominal values. If the frequencies are off, measurements made at the nominal values can show different readings because the peak will be at a different frequency. In some cases, noise or voice on the line can add to or subtract from these frequencies.

- **There is too much noise**: Noisy lines can introduce distortion to the digits. A simple inspection can identify if the lines have too much noise. If this is the case, Twist parameters should be increased and Ratio parameters should be decreased so that digits can be detected even when stronger noise and other signals exist.

- **Another signal is present that is similar to the DTMF digit**: Other signals present on the line, including some types of voice, can be detected as DTMF tones causing "talk-off." Adjusting the DTMF Detect and DTMF Delay parameters may correct this problem. However, changing the DTMF Digit Delay or DTMF Digit Detection timer values is a trade-off between improving DTMF detection and increasing the possibility of talk off. As the values of the timers are reduced, DTMF detection is improved, but the possibility of talk off increases. As the value is increased, the possibility of talk off is reduced, but the possibility of DTMF detection problems is increased.

You can adjust DTMF filter parameters to improve performance when the DTMF detection is not performing as desired. The filter parameters have one set of values when a recording is being played (Play Mode) and another set during all other functions (Idle Mode).

NOTE	If you change the DTMF parameters, the new parameters do not take effect until the voice processor is completely idle.

When digits are lost both in Play and Idle modes, the duration of the digits and the intervals between them may be too close to the DTMF Delay and DTMF Detection values. You should decrease these values to improve the chances that the digit characteristics will remain within acceptable values during the minimum required duration. Once the minimum on and off times are set according to the pattern of the digits being dialed, the next step is to verify the twist. A spectrum analyzer can show the amplitude for the frequencies in the signal, and the values

shown can be compared to the twist limits set in the driver. Rhetorex has a utility called "FFT" that will show the basic characteristics of the signal, including frequencies and amplitudes. Increasing the twist parameters for both Play and Idle modes will usually solve the problem.

When digits are lost only in Play mode, the Twist parameters should be increased and Ratio parameters should be decreased for Play mode only.

If talk-off is occurring, increasing the minimum duration of the digits should suffice. If this is not possible due to other constraints (for example., Speed Dial), Twist and Ratio parameters should be changed to make the filters less tolerant.

To program a parameter:

1. Select Voice Processor – **Timers and Limits**.

2. Double-click **DTMF Detection**.

3. Select the parameter. Parameter definitions begin on **page 935**.

4. In the **Value** column, select the new parameter from the list.

5. Press **ENTER** or click another field to save the change.

Parameter definitions are as follows:

- **DTMF Digit Detect On**: Determines the minimum duration of DTMF tones that can be recognized by Voice Processing during Play Mode; for example, when a prompt or message is playing. Valid settings for this parameter are 32, 48, 64, 80, or 96 milliseconds (msec). The default is 48 msec.

- **DTMF Digit Detect Off**: Determines the minimum silence allowed between DTMF tones that can be recognized by Voice Processing during Play Mode. Valid settings for this parameter are 32 or 48 msec. The default is 48 msec.

- **DTMF Digit Delay On**: Determines the minimum duration of DTMF tones that can be recognized by Voice Processing during Idle Mode; for example, when Voice Processing is waiting for input. Valid settings for this parameter are 32, 48, 64, 80, or 96 msec. The default is 32 msec.

- **DTMF Digit Delay Off**: Determines the minimum silence allowed between DTMF tones that can be recognized by Voice Processing during Idle Mode. Valid settings for this parameter are 32 or 48 msec. The default is 48 msec.

- **DTMF Digit Low To High Twist (Play and Idle)**: Reflects the difference between the DTMF digit high frequency and low frequency energy (high minus low). To allow the Voice Processor to detect a digit that consists of frequencies, the high frequency energy cannot be stronger than the low frequency energy by more than the setting of this parameter. In Play Mode the default for this parameter is 9.2dB. In Idle Mode the default is 8.2dB. Valid settings for this parameter are: No Limit, 6.7dB, 8.2dB, 9.2dB, 10.7dB, 12.7dB, 15.2dB, or 18.2dB.

- **DTMF Digit High To Low Twist (Play and Idle)**: Reflects the difference between the DTMF digit low frequency and high frequency energy (low minus high). To allow the Voice Processor to detect a digit that consists of frequencies, the low frequency energy cannot be stronger than the high frequency by more than the setting of this parameter. In Play Mode the default for this parameter is 8.2dB. In Idle Mode the default is 5.2dB. Valid settings for this parameter are: No Limit, 4.2dB, 5.2dB, 6.7dB, 8.2dB, 10.2dB, 13.2dB, or 18.2dB.

- **DTMF Digit In To In Ratio (Play and Idle)**: Indicates the digit energy minus the energy of the next digit. In order for the Voice Processor to detect a digit that consists of frequencies, the digit energy must be stronger than the energy of the next digit by the amount of this parameter setting. The default setting for Play Mode is 2.0dB. The default setting for Idle Mode is 8.0dB. Valid settings for this parameter are 1.0dB, 2.0dB, 3.0dB, 4.0dB, 6.0dB, 8.0dB, 9.0dB or 10.0dB.

- **DTMF Digit In To Out Ratio (Play and Idle)**: Indicates the digit energy minus the noise energy. For the Voice Processor to detect a digit that consists of frequencies, the digit energy must be stronger than the noise energy by at least the amount of this parameter setting. The default setting in Play Mode is 1.0dB. In Idle Mode, the default is 4.0dB. Valid settings for this parameter are 0.5dB, 1.0dB, 2.0dB, 3.0dB, 3.5dB, 4.0dB, 4.5dB, or 5.0dB.

- **DTMF Frequency Deviation**: This is the maximum variance from the standard frequencies allowed for valid DTMF. The range is 1.5%–2.5% (.1%); the default is 1.8%.

- **DTMF Maximum Valid Tone Dropout Time**: An otherwise valid tone may include this much silence and still be detected as valid DTMF. (Applies everywhere except during recording.) The range is 0–260 ms; the default is 15 ms.

- **DTMF Minimum Level Threshold**: This is the minimum per-frequency power for valid DTMF. The range is -14 dBm0 to -48dBm0; the default is -25.0 dBm0.

- **DTMF Minimum Valid Tone Off Time**: This is the minimum period of silence required between successive DTMF tones. The range is 20–260 ms; the default is 40 ms.

- **DTMF Minimum Valid Tone On Time**: This is the minimum length of a valid DTMF tone. The range is 20–260 ms; the default is 40 ms.

- **DTMF Negative Twist**: This is the maximum amount of negative twist allowed for valid DTMF. The range is 1 dB–16 dB; the default is 8.0 dB.

- **DTMF Positive Twist**: This is the maximum amount of positive twist allowed for valid DTMF. The range is 1 dB–16 dB; the default is 4.0 dB.

- **Recording - DTMF Minimum Valid Tone On Time**: This is an existing field. The field name changed from "DTMF Digit Detect for Recording - On (msec.)." The range is 20–260 ms; the default is 40 ms.

- **Recording - DTMF Minimum Valid Tone Off Time**: This is an existing field. The field name changed from "DTMF Digit Detect for Recording - Off (msec.)." The range is 20–260 ms; the default is 40 ms.

- **Recording - Maximum Valid Tone Dropout Time**: An otherwise valid tone may include this much silence and still be detected as valid DTMF. (Applies only during recording.) The range is 0–260 ms; the default is 15 ms.

DTMF GENERATION INFORMATION

The DTMF Generation folder contains the following information:

- **DTMF High Frequency Level**: This is the power level at which the high ("column") frequency component of DTMF tones is generated. The range is -63 to 0 dBm0; the default is -6 dBm0.

- **DTMF Low Frequency Level**: This is the power level at which the low ("row") frequency component of DTMF tones is generated. The range is -63 to 0 dBm0; the default is -6 dBm0.

- **DTMF Tone On Time**: This is the length of generated DTMF tones. The range is 40–260 ms; the default is 80 ms.

- **DTMF Tone Off Time**: This is the length of the silence period between successively generated DTMF tones. The range is 40–260 ms; the default is 80 ms.

To program DTMF Generation parameters:

1. Select Voice Processor – **Timers and Limits**.

2. Double-click **DTMF Generation Information** – <*option*>.

3. In the **Value** column, select the option from the list.

4. Click out of the field or press **ENTER** to save the change.

NUMBER OF VOICE CHANNELS

Applies to UVM voice processors only. Voice channels allow communication between the system and the applications. If a voice channel is not available in the application time slot group, callers to that application receive busy signals and the call recalls to the application attendant. At default, there are four available voice channels. The actual number of voice channels may be higher or lower depending on the number of Voice Processing circuit cards used in the external voice processing system and the port configuration of the cards.

The number of available voice channels determines the maximum number of channels that can be assigned to any single Time Slot Group. The combined total of channels assigned to Time Slot Groups may exceed the actual number of voice channels because it is unlikely that all time slot groups will use their maximum allotment at the same time.

NOTES	Although DB Programming allows you to configure 64 voice mail ports, Voice Processing using the Windows-based platform currently supports a maximum of only 32 ports (on the HX Base Server and PS-1 equipped platforms). Attempting to configure 64 voice mail ports may cause serious performance issues.
	This field is not programmable in online mode when connected to an external voice processor.
	Because UVM ports are licensable, customers upgrading their port capacity are required to purchase the corresponding licenses.

To set the number of voice channels (1–64):

1. Select Voice Processor – Timers and Limits – **Number of Voice Channels**.

2. In the **Value** column, click the current value, and then type the new value.

3. Click out of the field or press to **ENTER** save the change.

When using UVM, DB Programming displays the Number of Voice Channels field as read-only. This is because the number of channels and ports allocated for UVM are synonymous. Therefore, when you change the UVM port allocation field, the Number of Voice Channels field automatically updates with the port allocation value.

However, if you are using an external voice mail system, the Number of Voice Channels field remains writable and retains its previous functionality. After the UVM port resources are changed to 0, the number of voice channels and the maximum channel allocation fields for the time slot groups are no longer dependent on the value in the Number of Voice Channels field. By breaking this dependency, the system allows the customer to convert from a UVM system to an external type of voice mail system without having to reprogram time slot groups.

The range of values in the Number of Voice Channels field is 0–16. The default value is 4. The Number of Voice Channels field is programmable for UVM in offline mode but is read-only in online mode. This field is also *not* programmable in online mode when the MiVoice Office 250 is connected to an external voice processing system.

WINDOWS NETWORK LOGIN

The following information is used by the Voice Processor service when it logs on to the network. The Voice Processing Unit is required for these settings to take effect, and Unified Voice Messaging must be disabled. Refer to the *Voice Processing Unit Installation Manual*, part no. 835.2890, for details and information about setting up voice mail administrator rights for using the Avdap user account.

WINDOWS NETWORKING USERNAME

To change the username:

1. Click once on the current Value.

2. In the text box that appears, type the username that the Avdap service user account will use to log on to the network, then press **ENTER** or click another field to save the change.

WINDOWS NETWORKING DOMAIN

In this field, enter the name of the network domain in which the Voice Processing PC is a member.

To change the domain name:

1. Click once on the current Value.

2. In the text box that appears, type the name, then press **ENTER** or click another field to save the change.

WINDOWS NETWORKING PASSWORD

To specify the network password for the Avdap service user account:

1. Select Voice Processor – **Monitor Password**.

2. Right-click and select **Edit Password**. The Edit Monitor Password - Database Programming dialog box below appears.

3. Enter the current password, if one exists.

4. Enter the new password (up to 40 characters) in the New password box. The characters will not appear on the screen when typed; they will appear as asterisks (***).

5. Retype the password in the Confirm password box.

6. Click **OK** to exit and save the password. If the entered passwords match, you will return to the Password field. If not, you must re-enter the new password and verify it again. If you make a mistake while entering the password or want to leave it unchanged, select **Cancel**.

If Voice Processing is unable to log on to the network, a message is automatically entered into the Windows System Event Log. Since Voice Processing tries repeatedly to log in to restore a network connection, if the network is down for a long time (or Voice Processing is running in an environment without a network), many entries will be entered into the Event Log. Eventually, the log fills up and warning messages start appearing on the PC screen.

To change the setting of the Event Log:

1. From the Windows desktop, select Start – Programs – Administrative Tools (Common) – **Event Viewer**.

2. Choose **Log Settings** in the Log drop-down menu.

3. In the Event Log Settings screen, make sure that Event Log Wrapping is set to **Overwrite Events as Needed**.

4. Click **OK** to save the change and exit.

E-MAIL SYNCHRONIZATION WITH EM OPTIONS

In prior versions of DB Programming, the Voice Processor\E-Mail Synchronization (previously Unified Messaging) folder displayed a warning for systems with UVM stating that the feature was only available for Enterprise Messaging Voice Processors. This warning is now removed because E-Mail Synchronization is now supported by UVM, but the fields still do not apply to UVM and remain disabled as they pertain to faxing and other EM UM settings.

For complete information about Unified Messaging OSE for EM, refer to the latest version of the *Unified Messaging Open Standards Edition Administrator's Guide,* part number 835.3162.

NOTE	The existing Enterprise Messaging and Unified Messaging Open Standards Edition documentation have not been updated to apply the new UVM and E-Mail Synchronization terminology and will continue to use the existing Basic Voice Mail and Unified Messaging terminology throughout.

For programing E-mail Synchronization, see "E-mail Synchronization" on .

To program the E-mail Synchronization fields:

1. Select Voice Processor – **E-mail Synchronization**.

2. Configure the following fields:

   - **Attachment Format for Inbound Faxes**: Indicates the format of fax attachments in e-mails sent for Inbound faxes. The available options are TIF or PDF. It is set to TIF by default.

   - **Enable Delivery of Incomplete/Failed Inbound Faxes**: Enables e-mail notification for fax receivers regarding failed or incomplete inbound faxes. It is set to No by default.

   - **Enable IMAP Login for EM Server**: Enables remote IMAP logins to the EM server. It is set to No by default.

   - **EM Server IMAP Connection Timeout**: Determines the maximum duration that the Em server will wait for a remote IMAP connection to be established. The range is 30–1440 minutes; the default is set to 30 minutes.

3. Click out of the field or press **ENTER** to save the change.

E-MAIL GATEWAY

The following sections describe programming fields and procedures required to convert e-mail messages to voice mail messages with UVM systems.

For a complete explanation of the E-Mail Gateway feature, refer to the *Unified Messaging v2.3 Administrator's Guide* (part no. 835.3164), or *Unified Messaging Open Standards Edition Administrator's Guide* (part no. 835.3162).

Program the E-Mail System field first, as described below.

E-MAIL GATEWAY PROGRAMMING OPTIONS

The following sections describe E-Mail Gateway options and instructions. **Table 161** shows system features that use E-Mail Gateway fields. Empty fields may be programmed depending on the SMTP server configuration.

| NOTE | Remember to identify the E-Mail System first to determine the other fields that need to be programmed. |

Table 161: E-Mail Gateway Fields Used for System Features

SYSTEM FEATURE		FORWARD TO E-MAIL	VPIM	E-MAIL SYNCHRO-NIZATION	MOBILE MWI	CONFERENCE ACCESS CODE LIST
E-MAIL GATEWAY FIELD	**REF. PAGE**	**296**	**851**	**970**	**809**	**460**
Administrator E-Mail Address[1]	**943**	Recommended			Recommended	Recommended
E-Mail Address	**943**					
E-Mail Real Name[1]	**943**					
E-Mail SMTP Port	**944**					
E-Mail SMTP Server	**944**	Required for UVM	Required for UVM		Required	Required
E-Mail System	**944**	Required for UVM	Required for UVM	Required for UVM	Required	Required
E-Mail Username	**945**					
Gateway Password	**946**					

1. On the VPU, the "From" field on e-mail messages is based on what is programmed in the E-mail Real Name field. The "Mail From" is based on what is programmed in the Administrator E-Mail Address field. On the UVM, the E-mail Real Name is not used and both the "From" and "Mail From" fields are based on the Administrator E-Mail Address.

ADMINISTRATOR E-MAIL ADDRESS

This is the e-mail address of the system administrator. The system alerts the administrator of any problems sending the e-mail. This is the address in the "From" field in the e-mail and can be different than the E-mail Address (see **page 943**) used in the "Reply To" field.

On the VPU, the "Mail From" is based on what is programmed in the Administrator E-Mail Address field. On the UVM, both the "From" and "Mail From" fields are based on the Administrator E-Mail Address.

To enter the Administrator E-Mail Address:

1. Select System – E-Mail Gateway – **Administrator E-Mail Address**.

2. Click in the **Value** column, and then enter the address in the box. This field can contain up to 127 characters. Invalid characters are (: ; " \ | ( ) , < > ').

3. Click out of the field or press **ENTER** to save the change.

E-MAIL ADDRESS

The E-Mail Address option is the voice processing system e-mail address that is used in the "Reply To" field of an e-mail. This can be the same as the "From" field which is derived from the Administrator E-mail Address (see **page 942**). This address is only required if the E-mail System option (see **page 944**) is programmed to "SMTP."

NOTE	While this address can contain a "familiar" name, using SMTP, this should be a properly formatted E-mail address to avoid presenting a Reply-To e-mail address without a domain name and potentially being perceived as SPAM.

Below are examples of how an e-mail address is handled when the system using UVM has the Forward to E-mail feature enabled.

*   The Administrator E-mail Address and E-mail Address fields are the same: When the user receives the e-mail, the "From" field and the "Reply To" field show the same e-mail address.

*   The Administrator E-mail field is admin@test.com and the E-mail address field is chip@test.com: When the user receives the e-mail, the "From" field shows admin@test.com and the "Reply To" field shows chip@test.com

To enter the E-Mail Address:

1. Select System – E-Mail Gateway – **E-Mail Address**.

2. Click in the **Value** column, and then type the address in the box. This field can contain up to 127 characters. For example, this field might look like johndoe@mitel.com. Invalid characters are (: ; " \ | ( ) , < > ').

3. Click out of the field or press **ENTER** to save the change.

E-MAIL REAL NAME

The E-Mail Real Name specifies the voice processor's user name (such as VOICE MAIL). It is only programmable if the E-Mail System option is programmed to SMTP. When the voice mail computer sends an e-mail message, this name is included in the "From" field of the e-mail header.

On the VPU, the "From" field on e-mail messages is based on what is programmed in the E-mail Real Name field. On the UVM, the E-mail Real Name is not used and both the "From" and "Mail From" fields are based on the Administrator E-Mail Address.

To enter the E-Mail Real Name:

1. Select System – E-Mail Gateway – **E-Mail Real Name**.

2. Click in the **Value** column, and then type the address in the box. This field can contain up to 127 characters.

3. Click out of the field or press **ENTER** to save the change.

E-MAIL SMTP PORT

The E-mail SMTP Port is the port number for the Simple Mail Transfer Protocol (SMTP) server. It is only programmable if the E-Mail System option (see below) is programmed to "SMTP."

NOTE	This option applies to UVM only. If a MiVoice Office 250 is supported by an external voice messaging system, this option has no effect on E-mail Gateway functionality.

To change the E-Mail SMTP Port from the default 25:

1. Select System – E-Mail Gateway – **E-Mail SMTP Port**.

2. Click in the **Value** column, and then type the port number in the box. The range is 1–65535; the default is 25.

3. Click out of the field or press **ENTER** to save the change.

E-MAIL SMTP SERVER

The E-Mail SMTP Server option is the address of the SMTP mail server. It is programmable only if the E-Mail System option (see the following section) is set to "SMTP." The SMTP mail server is the server that the voice mail connects to send e-mail messages over the Internet. If this field is not set, the e-mail gateway features are disabled for the entire voice mail system.

To enter the SMTP Server:

1. Select System – E-Mail Gateway – **SMTP Server**.

2. Click in the **Value** column, and then type the server name in the box. The field can contain up to 127 characters.

NOTE	The format must be a fully qualified SMTP server name (not an IP address) to avoid DNS complications.

3. Click out of the field or press **ENTER** to save the change.

E-MAIL SYSTEM

The E-Mail System option specifies the type of e-mail system that is used to transfer messages. The value programmed in this option must correspond to your e-mail system. This field can be programmed to NONE or SMTP.

	To support VPIM networking (see **page 851**) or E-mail Synchronization (**page 970**), you must first set the E-mail System to "SMTP."
NOTES	If E-mail Gateway is set to SMTP and then changed to another setting, the following warning message appears: "If E-mail Synchronization is being used, the E-mail System is required to be SMTP. Changing the E-mail System will prevent E-mail Synchronization from occurring. Are you sure you want to make this change?" Click **Yes** to make this change, or click **No** to exit without making this change.

If the E-mail System is programmed to "SMTP," you must program the following fields:

- E-Mail SMTP Server on **page 944**

- E-Mail Address on **page 943**

- E-mail Username on **page 945**

- Gateway Password on **page 946**

- E-Mail Real Name on **page 943** (optional)

If it is programmed to "NONE," the voice processor E-Mail Gateway feature is disabled for the entire voice mail system.

To select the E-Mail System:

1. Select System – E-Mail Gateway – **E-Mail Pop Server**.

2. In the **Value** column, select either **None** or **SMTP**. The default value is NONE.

3. Click out of the field or press **ENTER** to save the change.

E-MAIL USERNAME

The E-Mail Username is the user name for the voice mail e-mail account on the SMTP Server. Before the voice mail computer can send or receive e-mail messages, it must log on to the underlying e-mail system. Depending on the SMTP configuration, this field may be required to log in to the SMTP server.

Therefore, the voice mail computer must have an account on the customer's e-mail system, and this field specifies the username for that account.

For authentication you only need to program a username and password. However, the Simple Mail Transfer Protocol (SMTP) mail software package supports three types of authentication (PLAIN, LOGIN, and CRAM-MD5) that happen automatically when it connects with the SMTP server. UVM supports PLAIN.

This package does not support Microsoft SPA (Secure Password Authentication), also known as NTLM. If the server supports CRAM-MD5, this is the most secure method supported.

To enter the E-Mail Username:

1. Select System – E-Mail Gateway – **E-Mail Pop Server**.

2. Click in the **Value** column, and then type the user name in the box. The field can contain up to 127 characters

3. Click out of the field or press **ENTER** to save the change.

GATEWAY PASSWORD

Optional. The Gateway Password is the password for the voice processing system e-mail account on the SMTP server. Before the voice processing system can send or receive e-mail messages, it must log on to the e-mail system. Therefore, the voice processing system must have an account on the customer's e-mail system.

NOTE	To provide system security, the e-mail system must have a password. To make the passwords difficult to guess, they should *not* consist of predictable patterns, such as one digit repeated several times.

To program the gateway password:

1. Select System – E-Mail Gateway – **Gateway Password**.

2. Right-click the **Password** field and select **Edit Password**. The Edit Gateway dialog box appears.

3. Type the current password, if one exists, then enter the new password in the **New password** box. Typed characters appear as asterisks (***). This field can contain up to 20 characters.

4. Retype the password in the **Confirm password** field.

5. Click **OK** to exit and save the password. If the entered passwords match, you return to the Password field. If not, you must re-enter the new password and verify it again.

E-MAIL GATEWAY FOR SYSTEMS EQUIPPED WITH A PS-1

If Dynamic Host Configuration Protocol (DHCP) is disabled, when the system is connected to UVM and the SMTP option is selected for the E-mail System field, the Hostname and the DNS fields in IP Settings are automatically checked. If these fields are not configured, you are prompted to configure them. The system then navigates you to the IP Settings folder.

The fields are associated with the Base Server.

Consequently, if you select the UVM and SMTP options on a system equipped with a PS-1, you are navigated to the System\IP Settings\Processing Server IP Settings folder instead of the System\IP Settings folder. On a system not equipped with a PS-1, you are navigated to System\IP Settings\Base Server IP Settings.

When the system is supported by UVM, two choices exist for the E-mail System option: None and SMTP. Only SMTP supports UVM. The following example shows where this option is located.
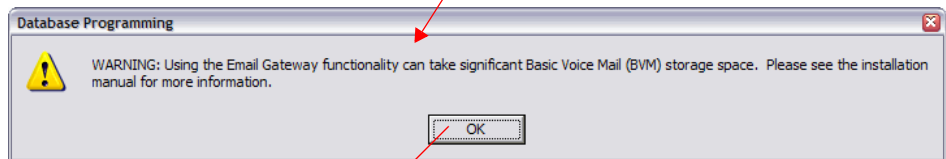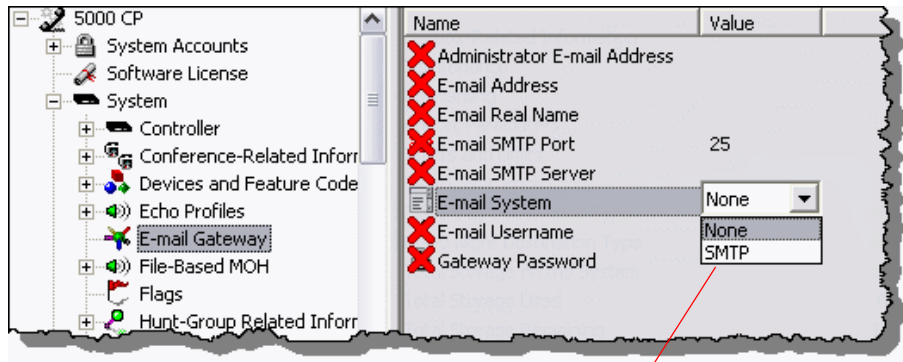
If not set:
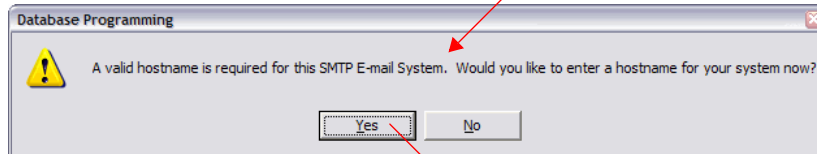
If not set:
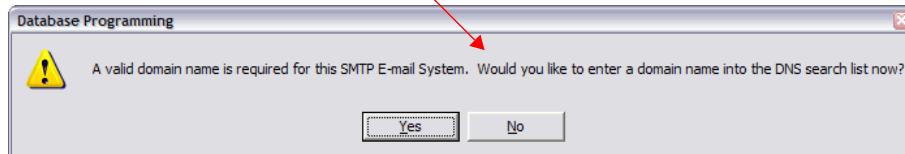
NOTE	Extra storage space is required when using the E-mail Gateway functionality. If using an external voice mail system, this prompt and message does not apply. The E-mail Gateway needs to be enabled for each mailbox per phone, thus reducing the overall UVM storage capacity.

The UVM e-mail system requires DNS and a valid hostname. Therefore, if DHCP is disabled you cannot delete the hostname or DNS Search List without causing the UVM e-mail system to fail. If UVM is being used and SMTP e-mail is configured, the configuration is prevented and a prompt appears.

- *If you click **Yes** in the prompt,* you are taken to the location mentioned in the prompt where you can configure the e-mail system options.

- *If you click **No**,* the attempt is denied.

To set up the e-mail gateway for SMTP:

1. Select System – E-Mail Gateway.

2. Select the **SMTP** option for the E-mail System. Click **OK** to the warning about storage space usage with UVM. Each mailbox configured to use this option uses additional memory. At a minimum, a 512 MB Mitel memory card is recommended.

The following fields are checked from the System\IP Settings\Base Server IP Settings or System\IP Settings\Processing Server IP Settings folder: hostname, DHCP flag, and DNS Search List.

For SMTP to work with UVM, a complete hostname and domain name must be provided for the system. If DHCP is enabled, only the hostname is required. If DHCP is disabled, the DNS search list must contain a valid domain name for SMTP to work. As indicated in the example on **page 947**, you will be prompted for this information. Without this information, you cannot select the SMTP option. Click **Yes** to automatically navigate to the IP Settings folder where you can set the appropriate DNS options. Also, if the hostname is configured, but DHCP is disabled and the DNS Search List is blank, you are prompted to set this information.
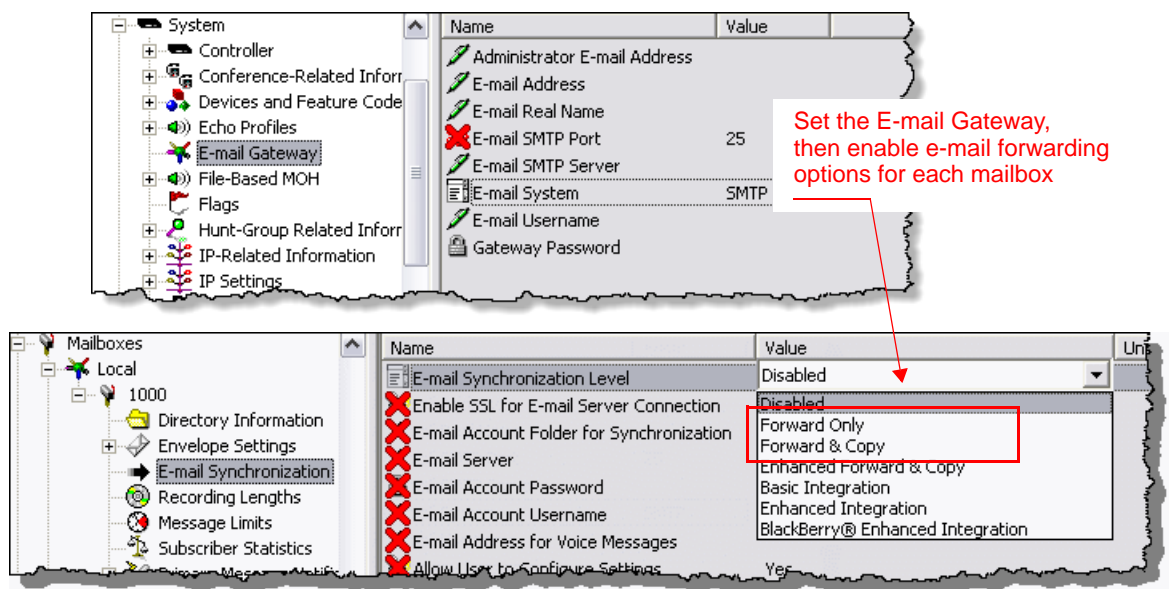
3. Navigate to each mailbox for which you want to enable e-mail forwarding options, and enable the E-Mail Gateway settings from the **E-mail Synchronization** folder. Select from the following options. For details, refer to the *MiVoice Office 250 DB Programming Help*.

| NOTE | The options that are not listed below are not supported by UVM. |

- **Forward Only**: All voice mail messages delivered to the mailbox are forwarded to the e-mail address specified in the mailbox e-mail Address field. In this configuration, voice mail messages are not saved in the mailbox. When the user deletes the e-mail message containing the voice mail message, or when e-mail delivery fails for any reason, all record of the voice mail message disappears. The following prompt appears upon selecting the Forward Only option.
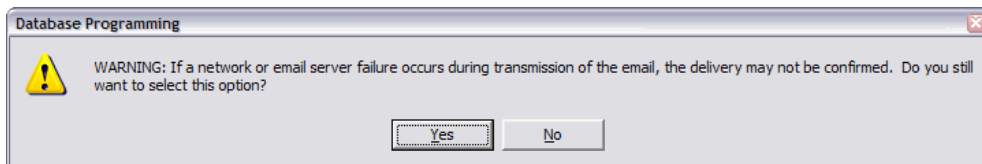
- **Forward and Copy**: All voice mail messages for the mailbox are delivered to the mailbox and a copy is forwarded to the mailbox e-mail Address. If one is deleted, the other is not affected.

| NOTE | If both Remote Messaging and Forward to E-Mail features are enabled, you must select the Forward and Copy option. If the Forward Only option is selected, the user does not receive Remote Messaging notifications. |

FAX-ON-DEMAND

The Fax-On-Demand feature provides fax services to callers. It is a specially programmed Call Routing Announcement application that uses digit translation to allow callers to select the documents they want to have faxed to them. Fax-On-Demand is available in external voice processing systems only.

With Fax-On-Demand, callers can use a DTMF phone to request one or more documents from the company's fax library. When the request is completed, the voice processor places a call to the caller's fax machine to deliver the requested documents.

To use Fax-On-Demand the following fields must be programmed:

- Timers and Limits below

- Fax Documents on **page 952**

- Allow International Calls on **page 953**

- Outgoing Access on **page 953** (trunk group)

  - Outgoing Access

  - Outgoing Access Prefix

  - Outgoing Access Termination

- Start/Stop Time on **page 954**

- Days of the Week on **page 954**

- Fax Format on **page 954**

  - Format - Local Fax ID

  - Format - Company Name

  - Format - Logo Document

FAX-ON-DEMAND TIMERS AND LIMITS

To view the list of current values, double-click **Timers and Limits**.

To program a timer or limit:

1. Select Voice Processor – **Fax-On-Demand**.

2. Double-click **TImers and Limits**.

3. In the **Value** column, click the current value, and then type the new setting in the box.

4. Press **ENTER** or click another field to save the change.

The following timers and limits can be programmed for the fax feature:

- **Fax Retry Timer**: When the voice processor is unable to complete a fax delivery because the line is busy or there is no answer, it will wait until this timer expires to attempt the delivery again. The range for this field is 1–255 minutes. Default is 10 minutes.

- **Fax Retransmission Timer**: When the voice processor is unable to complete a fax delivery because the connection failed (for example, the receiving fax machine had a power failure), it will wait until this timer expires to attempt the delivery again. The range for this field is 1–255 minutes. Default is 1 minute.

- **Maximum Fax Delivery Attempts**: This is the number of times the voice processor will attempt to send a fax when the number is busy, there is no answer, or there are transmission errors. If the system encounters unavailable resources (fax ports, documents, outgoing calls, or outgoing trunks) the attempt does not count toward the Maximum Fax Delivery Attempts. The allowed range for this field is 1–15 attempts. It defaults to 5 attempts.

- **Maximum Fax-On-Demand Ports**: This sets the maximum number of fax ports the system can use for performing Fax-On-Demand (either delivering outgoing faxes or importing fax documents from the system administrator mailbox). By placing a limit on the number of Fax-On-Demand ports, you can reserve fax ports for receiving incoming faxes through mailboxes and Call Routing Announcement applications. For example, if the system has eight fax ports and the Maximum Fax-On-Demand Ports field is set to six, there will always be at least two ports available for faxes received through mailboxes and Call Routing and six ports for outgoing faxes. If the Maximum Fax-On-Demand Ports field is programmed to a number that exceeds the actual fax ports available, the software will automatically adjust the limit.

- **Automatic Header Reduction**: This tells the voice processor how much of each document you fax into the system must be removed to erase the sender information at the top of the document. If the fax machine you use to enter the documents does not place sender information at the top of the document, you can set this field to 0. You can reduce the header 0–160 sixteenths of an inch (0–10 inches). This field defaults to 4-sixteenths of an inch (0.25 in.).

- **Fax Tone Wait Timer**: This is the amount of time the voice processor will wait for fax tone before sending or receiving a document. If it does not receive fax tone before this timer expires, it will hang up. If the system was sending a fax, it will attempt the call again after the Fax Retry Timer expires. The range for this field is 1–255 seconds. Default is 40 seconds.

- **Maximum Fax Selections**: This determines the number of faxes a caller can select at a time. When the caller has selected the maximum number of documents, the voice processor prompts them for their fax number. After entering the fax delivery information, the caller can then go through the process again to request more documents without hanging up and calling in again. However, each series of requests generates a separate outgoing call from the voice processor, each time the fax delivery information is entered by the caller. The allowed range for this field is 1–20 documents. It defaults to 10.

- **Maximum Fax-On-Demand Library Size**: This determines the amount of voice processor hard disk space that will be allotted for storing fax documents. The allowed range for this field is 0–255 megabytes; however, the actual maximum depends on the available disk space on the voice processor hard drive. It defaults to 0 megabytes and must be programmed to a higher value before any fax documents can be imported. If the library size is reduced while there are documents stored in the library, the reduction will not affect existing documents, even if they exceed the new maximum. However, new documents cannot be added unless there is sufficient disk space available.

- **Current Fax-On-Demand Library Size And Percentage Of Maximum Currently Used**: This shows the amount of disk space that is currently being used by the documents in the

fax library and the percentage of the Maximum Fax Library Size that has been used. These fields cannot be programmed. They are shown for reference only.

FAX DOCUMENTS

To program Fax Documents:

1. Double-click **Fax Documents**, the documents currently existing in the system are shown in the list.

2. Program the number and description of a document as follows:

    - **Document Number**:

        a. Type a number, up to four digits, to identify the fax document in the text box.

        b. Press **ENTER** or click another field to save the change. This number will be used by programmers to select the destination for digit translations and by callers when selecting fax documents by number.

    - **Description**: This is the abbreviated description that is used in programming screens. It can have up to 20 characters.

        a. Enter the description in the text box.

        b. Press **ENTER** or click another field to save the change.

To see additional fields: Double-click a document to see these additional programming fields:

- **Detailed Description**: This is the description that appears on the fax cover sheet and in any fax programming reports. It can include up to 40 characters.

    a. Enter the description in the text box.

    b. Press **ENTER** or click another field to save the change. If you do not enter a detailed description for the document before exiting, a warning message appears.

- **Statistics:** This information cannot be programmed. It is shown as a reference to indicate the following:

    - **Number of Requests**: How many times the document has been requested, since statistics were last cleared (see **page 1027**).

    - **Last Request**: The last date that the document was requested, since statistics were last cleared. This includes requests that could not be delivered because of transmission errors.

    - **Last Modification**: The date and time of the last modification to the document being programmed, since statistics were last cleared. Or, if the document has not been modified, it shows the time that it was imported.

    - **Pages**: The number of pages included in the document.

    - **Image File Size**: The amount of disk space occupied by this document.

ALLOW INTERNATIONAL CALLS

This field allows you to enable or disable international fax deliveries. If it is disabled, the prompt asks the caller to enter a 10-digit number for their fax destination. If this field is enabled, the prompt gives international dialing instructions and allows international and domestic numbers. The field defaults to disabled.

To allow international calls:

1. Select the check box to place a check mark in it to change the setting to **Yes**.

2. Press **ENTER** or click another field to save the change. To set it back to **No**, click the check box again.

OUTGOING ACCESS

The next three fields control the outgoing access for calls placed by the Fax-On-Demand application.

Outgoing Access: This is the outgoing trunk access code that the Message Notification/ Retrieval application will use to place the outgoing fax delivery call. The default value is blank. To determine the outgoing access trunk group, use one of the following methods:

Method A

1. Select the current Value, and then enter the new value in the text box.

2. Press **ENTER**. A dialog box appears displaying what is associated with the number entered.

3. Click **OK**. The new number appears in the field.

Method B

1. Right-click the existing value and select **Change Outgoing Access**. A dialog box appears prompting for the device type to include.

2. Select **CO Trunk Group**, and then click **Next**. The list of trunk groups appears. You can view them in a list by selecting the List button or view details by selecting the Details button.

3. Select the appropriate trunk group, and then click **Finish**. The selection appears in the Outgoing Access field.

Outgoing Access Prefix: This is the digit dialed by the voice processor before the outgoing access code, if needed (for example, a forced account code). This field can include up to 18 characters. Valid characters include digits 0–9, * and #, and "P" for pause. This field is blank by default. To enter the prefix:

1. Click the **Value**, and then enter the address in the text box.

2. Press **ENTER** or click another field to save the change.

Outgoing Access Termination: This is the digit dialed by the voice processor to terminate the outgoing access code, if needed. This field can include up to 18 characters. Valid characters include digits 0–9, * and #, and "P" for pause. This field defaults to #. To enter the termination digit:

1. Click the Value, and then enter the address in the text box.

2. Press **ENTER** or click another field to save the change.

START/STOP TIME

These fields determine the time period during which faxes will be sent to callers. If a caller requests a fax after the Stop Time, the system will not send the fax until the Start Time. To provide 24-hour fax service, set both fields to the same value. Both fields default to 5:00 PM.

To set the time:

1. Select the current Value, then select the hours, minutes, or AM/PM field, and use the arrows to scroll to the desired setting.

2. Select another field or press **ENTER** to save the change.

NOTE	With a U.S. system, the Fax-On-Demand feature uses User Group 1 and the Home Area Code for toll restriction information. Ensure that these are programmed correctly for outgoing fax delivery calls. Also ensure that the Message Notification/Retrieval application has the appropriate trunk access.

DAYS OF THE WEEK

These fields set the days of the week during which faxes will be sent to callers. If a caller requests a fax on a disabled day, the system will wait for the next enabled day to send the fax. At default, all days of the week are enabled. You cannot disable all seven days; at least one must be selected.

To enable a specific day of the week:

1. Select the check box to place a check mark in it and change the setting to **Yes**.

2. Press **ENTER** or click another field to save the change. To set it back to **No**, select the check box again.

FAX FORMAT

The following fields affect the format of the fax documents that are sent by the Fax-On-Demand application.

- **Local Fax ID**: The Local Fax ID appears at the top of each page sent by the voice processor. The ID should include the number of the voice processor and/or the company name. The ID can be up to 20 characters. Valid characters include upper-case letters, digits 0–9, and the plus sign (+).

  To enter the Fax ID:

  a. Click the **Value** field, then enter the address in the text box.

    b. Press **ENTER** or click another field to save the change.

- **Company Name**: The system places Company Name on the "From" line of the fax cover sheet. The company name can include up to 40 characters. Any character is allowed.

    To enter the company name:

    a. Click the **Value** field, then enter the address in the text box.

    b. Press **ENTER** or click another field to save the change.

Logo Document: After you have imported your company logo, you can assign that fax document as the Logo Document for your fax cover sheets. For more information about importing fax documents, refer to the *MiVoice Office 250 Unified Voice Messaging Administrator Guide,* part number 580.8009. Any fax document in the database can be designated as the logo document. The logo can be up to 5.5 inches tall. If desired, you can choose not to have a logo on your cover sheets by selecting **None**. To indicate which fax document is the logo document, use one of the following methods:

Method A

1. Select the current Value, then enter the new value in the text box.

2. Press **ENTER**. A dialog box appears displaying what is associated with the number entered.

3. Click **OK**. The new number appears in the field.

Method B

1. Right-click the existing value and select **Change Format**, and then select **Logo Document**. A dialog box appears prompting for the device type to include.

2. Select **Fax Document** (or select **None Device**, if you do not want to use a logo document), and then click **Next**. The list of currently existing documents appears. You can view them in a list by selecting the List button or view details by selecting the Details button.

3. Select the appropriate document (or None), and then click Finish. The selection appears in the Logo Document field.

SMDR BUFFER

The voice processor buffer SMDR records over an IP connection from the local node and records that are sent from other nodes in the network. However, only the voice processor or other system connected to the network can receive buffered SMDR. Buffered SMDR is ***not*** provided in UVM. A message appears at the bottom of the DB Programming screen status area if trying to configure these options with UVM. The following information can be programmed for the voice processor SMDR Buffering application.

- Clear SMDR Buffer

- SMDR Buffer Size

- SMDR Buffer-Box Bit Rate - COM 2

- Enable SMDR Buffering

Clear the SMDR buffer: *(Not available in stand-alone programming sessions.)* During a direct-connect or remote programming session, you can clear the data in the SMDR buffer.

To clear the SMDR buffer:

1. Right-click **SMDR Buffer** in the Voice Processor directory and select **Clear**. A dialog box appears prompting for confirmation to clear the SMDR buffer.

2. Select **Yes** to continue or select **Cancel** to leave the buffer unchanged. If you chose to clear, an information window appears informing you that the buffer has been cleared. Click **OK** to continue.

Programming the SMDR Buffer: You can program the following fields:

- **SMDR Buffer Size**: To determine how much of the Voice Processing PC disk space will be devoted to SMDR buffering:

  a. Select the current value, then enter or scroll to the desired number of megabytes (0–20).

  b. Press **ENTER** or click another field to save the change.

    If you attempt to set the buffer size to a value higher than your available disk space, a warning message appears that displays, "The Voice Processing PC did not have enough disk space to set the buffer size to XXX megabytes. The current buffer size is now 0 megabytes. Note: If there was data in the buffer, the data is still intact." Click **OK** to continue and reset the buffer to a valid size.

    If you change the size of the buffer, a dialog box appears informing you that the buffer will be cleared and prompts for confirmation to continue. Click **Yes** to complete the resizing or click **No** to leave the buffer and buffer size unchanged.

- **SMDR Buffer Box Bit Rate**: To determine the rate of output from the buffer box port (COM2):

  a. Select this drop-down list box. The available rates are 300, 1200, 2400, 4800, and 9600.

  b. Press **ENTER** or click another field to save the change.

- **Enable SMDR Buffering**: To enable the SMDR Buffering option:

    a. Select the check box to place a check mark in it.

    b. Press **ENTER** or click another field to save the change. To disable SMDR buffering, select the check box again to remove the check mark.

Chapter 15

SUBSCRIBER MAILBOXES

INTRODUCTION

This chapter provides information to create and configure subscriber mailboxes for Unified Voice Messaging (UVM).

For MiCollab Unified Messaging system setup and mailbox configurations, refer to the following resources:

- *MiVoice Office 250 and MiCollab Unified Messaging Integration Guide*

- MiCollab Unified Messaging System Administration Help

- MiCollab Unified Messaging Technician's Handbook

- *MiVoice Office 250 Installation Manual*

- MiVoice Office 250 DB Programming Help

Program the voice processor system settings before creating and programming mailboxes. See "Voice Processor Features and Programming" on **page 835**.

A mailbox is a storage location that stores all messages that have been directed to it (including prompts, greetings, and special programming.) Each "Subscriber" (member of the voice message system) is assigned a unique mailbox number. At system default, the passwords are the same as the mailbox numbers. (For example, the default password for mailbox 1001 is "1001.")

Voice mailboxes are stored on specifically formatted compact flash-type memory card. For more information about flash memory, refer to the Specifications and Product Description chapters in the *MiVoice Office 250 Installation Manual* . For storage capacity beyond what is available with UVM, use an external voice processing system, for example UVM or MiCollab Unified Messaging.

NOTICE

Possible Memory Card Corruption. Remember the following when using flash-type memory cards:

- Do *not* remove or install the compact flash-type memory card while the system is up and running or power is otherwise supplied to the Base Server, as this may damage the memory card. Shut down the system using the LCD panel and unplug the power cord before removing or inserting the memory card.
- Using any other compact flash-type memory card than that provided by Mitel is not supported.
- Do *not* place the compact flash-type memory into any other computer. Doing so may corrupt the file system and software.

The following mailbox types of mailboxes are available for UVM:

- **Standard mailbox**: Subscribers have all of the available options as programmed. Standard mailbox features include:
    - Listening to new messages
    - Recording and sending messages
    - Deleting and restoring messages
    - Saving messages

- • Accessing personal options (greeting, password, and so forth)

- • Accessing remote messaging (if authorized through database programming)

- • **Receive-Only mailbox**: Subscribers may only listen to new messages, save, delete, or recover them and access personal options. This type of mailbox is restricted from recording and sending messages and has no access to remote messaging.

- • **Voice Mail Administrator**: The Voice Mail Administrator has all of the options a standard mailbox subscriber has, plus the following features:

  - • Record and send broadcast messages to all subscribers of the Voice Mail system

  - • Provide mailbox and group list maintenance

  - • Record and select custom Audiotex recordings

- • **UVM Play-Only Mailbox**: *Applies to UVM unassociated mailboxes only.* The UVM "play-only" mailbox option prevents the caller from leaving a message. The "play-only" mailbox is similar to an "announcement-only" application where it plays the designated greeting once and then the call automatically disconnects after the greeting finishes playing. This feature allows a subscriber to record a custom message without requiring administrative access to voice mail. For example, a teacher (the "subscriber") can record a homework assignment message for students who missed class that day to retrieve. Each UVM mailbox in DB Programming, has an option to indicate if it is "play-only." The default value is set to "No" for all mailbox types.

- • **Extension IDs**: Extension IDs are used in conjunction with transferring calls through the Automated Attendant or using the Extension Directory. The extension ID allows callers to be transferred to phone extensions which do not have mailboxes and allows the phone user to record a directory name and set up a password.

Calls transferred from the Automated Attendant or a CRA application to a phone with a mailbox can be screened, announced, or unannounced. See "Extension IDs" on **page 920**.

During database programming, each mailbox is assigned a dial-0 operator destination. When a caller presses **0** while listening to the mailbox personal greeting or recording instructions, or while recording a non-subscriber message, the operator destination is called. The operator can be one of the following types:

- • **Mailbox:** If the operator is a mailbox, the caller hears the operator destination personal greeting after pressing **0**.

- • **Phone extension number:** If the operator destination is an extension number, the caller hears the Automated Attendant transfer prompt while the call is placed to the extension number ("*Please hold while I transfer your call to..."*), unless it has been disabled at the mailbox. If the phone user does not answer, the caller is prompted to leave a message, if the operator's extension has an associated mailbox. If the caller chooses not to leave a message, the call returns to the main menu of the application being used (CRA, Voice Mail, or Auto Attendant).

- • **Application extension number:** If the operator destination is an application extension number, the call is transferred to that application main menu.

- • **Operator:** If the operator destination is set to "Operator," the caller is transferred to the operator destination programmed in the Voice Processing database.

- • **None:** If the operator destination is set to "None," the **0** is ignored.

CREATING MAILBOXES

The following sections describe how to create and program mailboxes for nodes. A mailbox is a storage location on the voice processing computer hard disk that stores all messages that have been directed to it. Mailboxes can be either *associated* or *non-associated:*

- **Associated**: A mailbox that is directly associated to an extension number.

- **Non-Associated**: A mailbox that has an extension number that does not match the phone extension number. For example, a hunt group extension number can have a mailbox, but the system sends message indications to a designated hunt group phone, which uses a different extension number than that of the hunt group mailbox.

NOTE	If you are using non-associated mailboxes, you must disable the Validate Voice Mailbox flag. When enabled, this flag prevents users from dialing mailbox numbers that do not match valid extension numbers.

You can create associated and non-associated mailboxes on the local node. However, you cannot create non-associated mailboxes for phones on other system nodes. For more information, see "Mailbox-Related Information" on **page 995**.

When you view system mailboxes, the nodes that are shown are determined by whether you have previously programmed Voice Processing network nodes. If nodes exist, you will see the nodes listed with the local node. You can click any of them to program the mailboxes for the selected node, as described below. If there are no voice processing networking nodes, you skip directly to the list that shows mailboxes on the local node.

If you are using Enhanced Integration or Blackberry Enhanced Integration for UVM E-mail Synchronization, you can program up to 100 (for a system without a PS-1) and 250 (for a system with a PS-1) mailboxes. If you attempt to program more than 250 mailboxes for E-mail Synchronization, a warning message appears and DB Programming cancels the changes.

See **page 970** for details about E-mail Synchronization.

This section contains the following programming instructions:

- **Creating Mailboxes for Extensions with Extension IDs** below

- Creating Associated Mailboxes on **page 963**

- Creating Non-Associated Mailboxes on **page 963**

- Changing Non-Associated Mailbox Extension Numbers on **page 964**

- Deleting Mailboxes on **page 964**

- Clearing Mailbox Messages on **page 964**

- Copying Mailbox Settings on **page 965**

CREATING MAILBOXES FOR EXTENSIONS WITH EXTENSION IDS

You can create mailboxes for extensions that currently have extension IDs.

To create a mailbox for an extension that currently has an extension ID:

1. Select Voice Processor – Devices – Mailboxes – <***node***>.

2. Delete the extension ID.

3. Create the mailbox.

CREATING ASSOCIATED MAILBOXES

You can create associated mailboxes for either local-node or off-node systems.

To create an associated mailbox:

1. Select Voice Processor – Devices – Mailboxes – <***node***>.

2. Right-click in the right pane, and then click **Create Associated Mailbox**. A window appears prompting for the device type to include.

3. Select the device type, and then click **Next**. The list of devices with details appears. To view items in a list only, click **List**.

4. Select the devices for which you want to create mailboxes (you can press SHIFT or CTRL to select more than one device), and then click **Add Items**. The selections appear in the mailbox list.

5. Click **Finish** to exit.

CREATING NON-ASSOCIATED MAILBOXES

You can create non-associated mailboxes for local nodes. You cannot create non-associated mailboxes for phones on other system nodes. For more information, see "Mailbox-Related Information" on .

To create a non-associated mailbox:

1. Select Voice Processor – Devices – Mailboxes – <***node***>.

2. Right-click in the right pane, and then select **Create Non-Associated Mailbox**. The Create Mailbox Extension dialog box appears.

3. In the **Starting Extension** box, enter the starting extension for the new mailboxes.

4. In the **Number of Extension**s box, enter the number of extensions.

5. Click **OK**.

6. The new mailboxes appear in the right pane. If any mailbox extensions conflict with existing extensions, a dialog box appears, listing the mailboxes that were not created.

7. Program the description and user name as follows:

   a. Enter a description of up to 20 characters in the **Description** box. This description is used in the mailbox directory and should be entered in the form "last name, first name" (with a comma and space separating the names).

   b. Enter a name of up to 10 characters in the **Username** box. This is the name that appears on phone displays. Do *not* use slash (/), backslash (\), vertical slash (|) or tilde (~) characters in usernames. Do *not* use Control characters in descriptions or usernames.

CHANGING NON-ASSOCIATED MAILBOX EXTENSION NUMBERS

You can change non-associated mailbox extension numbers.

To change a non-associated mailbox number:

1. Select Voice Processor – Devices – Mailboxes – *<node>* – *<mailbox number>*.

2. In the **Extension** column, select the number, and then type or select the new number.

3. Click out of the field or press **ENTER** to save the change.

DELETING MAILBOXES

You can delete associated or non-associated mailboxes from local and remote nodes.

> **NOTICE**
>
> **Disruption of Voice Processing Possible.** Make sure a mailbox is not in use before deleting it. Deleting a mailbox while it is in use causes serious performance issues for voice processing.

To delete mailboxes:

1. Select Voice Processor – Devices – Mailboxes – *<node>*.

2. Select the mailboxes.

3. Right-click, and then select **Delete**. The mailboxes are removed from the list.

CLEARING MAILBOX MESSAGES

If you have an active direct connection or remote programming session, you can clear (erase) all of the messages in a mailbox.

To clear mailbox messages:

1. Select Voice Processor – Devices – Mailboxes – *<node>* – *<mailbox number>*.

2. Select the mailbox(es) you want to clear.

3. Right-click, and then select **Clear Message(s)** from the Selected Mailboxes option.

COPYING MAILBOX SETTINGS

To save time, you can copy mailbox settings and paste the settings in another mailboxes.

To copy and paste mailbox settings:

1. Select Voice Processor – Devices – Mailboxes – *<**node**>*.

2. Right-click the mailboxes, and then select **Copy**.

3. Right-click the mailbox where you want to copy the settings, and select **Paste**. A dialog box appears, as shown below, that allows you to select the attributes you want to copy.
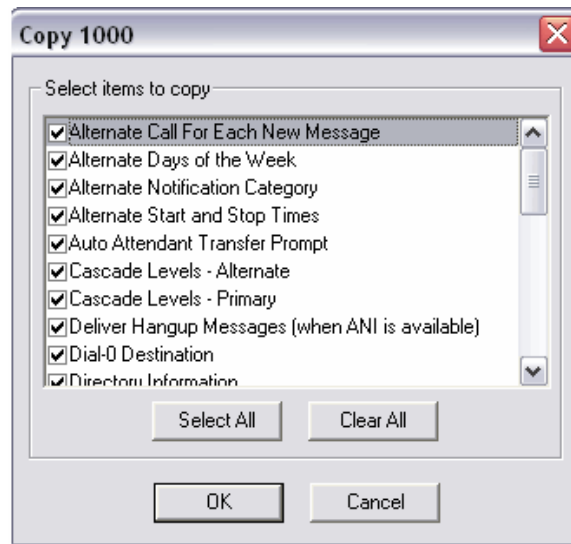
4. Select the options that you want paste, and then click **OK** to save your changes.

NETWORK MAILBOXES (OFF-NODE MAILBOXES)

> **NOTICE**
>
> **Unstable System Performance Possible.** Do not create or delete more than 2000 off-node devices at a time. Batch-creating more than 2000 off-node devices may cause system instability or other problems.

Mailboxes that are programmed on the local node but are associated with mailboxes on remote voice processing nodes are referred to as "network mailboxes." They are used by the local node to identify and locate the mailboxes located on the other nodes. They are not actual mailboxes; they are just "place holders" that tell the local node where to send messages received for that mailbox number. If the mailbox subscriber logs on to a network mailbox, the options available are the directory name, the greetings, the password, and possibly the transfer method programming prompt, if enabled. In addition, if a subscriber changes the password, directory name, or greeting selection, these changes will be automatically updated on the corresponding network mailboxes.

You can program the following options only for off-node mailboxes:

- **Remote Mailbox Extension** below

- Unlisted Number on **page 969**

- Private Extension and Mailbox on **page 969**

- Allow Transfer Method Programming on **page 988**

- Auto Attendant Transfer Prompt on **page 988**

- Passcode on **page 990**

- Transfer Method on **page 991**

REMOTE MAILBOX EXTENSION

The Remote Mailbox Extension is for all off-node mailboxes, associated and non-associated. The Remote Mailbox Extension is the mailbox number that is sent in the VPIM message to the remote node. This mailbox extension number should match the destination mailbox on the remote VPIM system.

The Remote Mailbox Extension allows the system to send the same or different mailbox number to the remote system. For example: VPIM message, 3000.Node3-5000.mycompany.com. The local MiVoice Office 250 has a non-associated mailbox 4200, but the Remote Mailbox Extension for this mailbox is set to 3000, so the system will send VPIM messages to mailbox 3000 on Node 3. When the local user records and sends messages to 4200, the system sends 3000 in the VPIM message.

> **NOTE** This is the only field on the MiVoice Office 250 UVM that allows you to modify the mailbox digits. There are no other digit modification tables on the MiVoice Office 250 UVM systems.

The following are some considerations:

- The Remote Mailbox Extension is populated with the matching associated mailbox extension number. The Remote Mailbox Extension can be changed to another value if necessary to send to another mailbox. You can have several mailboxes on one location send VPIM messages to the same mailbox on the remote location. Typically, the associated off-node

mailboxes would have the same Remote Mailbox Extension number, but this is not validated where it can provide flexibility.

- The Remote Mailbox Extension is left blank and must be manually programmed when non-associated mailboxes are created. Type the necessary mailbox number to which to route on the remote system.

- You can change the Remote Mailbox Extension on associated and non-associated mailboxes.

- The Remote Mailbox Extension can support up to 5-digit mailboxes.

If there is a universal numbering plan used by all of the networked voice mail nodes, the Remote Mailbox Extension option is the same as the local mailbox number. If the numbering plan is *not* universal, the Remote Mailbox Extension option can be different than the local mailbox number. The mailbox numbers programmed in this option are not verified because the local voice processing system does not know about the mailbox numbers on the remote voice processing system until it actually attempts to deliver a message to the remote voice processing system. The default value for the Remote Mailbox Extension is the local mailbox number.

The Remote Mailbox Extension option can be changed to any number up to five digits. The remote mailbox option is not validated and it should *not* be programmed a part of a group list or network mailbox on the remote node.

UNLISTED NUMBER AND PRIVATE MAILBOX NUMBER OPTIONS

A network mailbox should have the Unlisted Number and Private Mailbox Number options enabled. The Unlisted Number option keeps the network mailbox from appearing in the company directory, and the Private Mailbox Number option prevents the caller from hearing the mailbox number while using the company directory. These two fields can only be programmed from DB Programming.

PROGRAMMING MAILBOX OPTIONS

NOTE	Mailboxes that are programmed on a local node but are associated with mailboxes on remote voice processing nodes are referred to as "network mailboxes." Network mailboxes have limited programming options. See "Network Mailboxes (Off-Node Mailboxes)" on **page 966** for more information.

You can program the following mailbox options:

- Directory Information on **page 969**

- Envelope Settings on **page 969**

- E-mail Synchronization on **page 970**

- Recording Length on **page 976**

- Message Limits on **page 976**

- Subscriber Statistics on **page 977**

- Dial-0 Destinations on **page 978**

- Remote Messaging on **page 979**

- Primary and Alternate Message Notification on **page 980**

- Receive Only on **page 987**

- Allow Transfer Method Programming on **page 988**

- Play Recording Instructions on **page 988**

- Auto Attendant Transfer Prompt on **page 988**

- Deliver Hangup Message (when ANI is available) on **page 989**

- Swap "7 for Save" and "9 for Delete" Message Keys on **page 989**

- Designate this Mailbox for Play Only on **page 990**

- Passcode on **page 990**

- Greeting on **page 991**

- Transfer Method on **page 991**

- Primary and Alternate Message Notification on **page 980**

- Message Notification Phone on **page 992**

- Time Zone on **page 993**

- Quota Warning on **page 993**

- Quota Grace on **page 993**

- Language on **page 994**

DIRECTORY INFORMATION

Mailbox Directory Information options include the following:

- **Unlisted number**: The mailbox number is not included in the mailbox directory, but callers can dial the number.

- **Private extension and mailbox numbers**: The extension *and* mailbox numbers are not included in directory, but callers can dial either number.

By default, mailbox and extension numbers are not either private or unlisted.

To set Directory Information flags:

1. Select Voice Processor – Devices – Mailboxes – *<node>* – *<**extension**>*.

2. Double-click **Directory Information**. The list of flags appears.

3. Select the option that you want to change.

4. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

5. Click out of the field or press **ENTER** to save the change.

ENVELOPE SETTINGS

E-mail, fax, and voice messages can include a "message envelope" for all message types, including those that are converted to speech. This envelope contains information such as the message duration, source, and so forth. Announcement options include the following:

- **Announce Message Length**: (*Voice Only*) If enabled, the voice mail message duration is included in the message envelope.

- **Announce Message Subject**: (*E-mail Only*) If enabled, the subject line of the e-mail message is included in the envelope. If the subject line is blank, the envelope ignores this setting and does not announce the subject.

- **Announce Message Pages**: (*Fax Only*) If enabled, the total number of pages that were faxed are included in the envelope.

- **Announce Message Source**: If enabled, the envelope includes the message originator. For e-mail messages, this is the e-mail address or alias of the message sender; for voice mail messages, this is the name and number of the caller (if available); and for faxes, this is the originating fax number.

- **Announce Date and Time**: If enabled, the date and time that the message was received is included in the envelope.

You cannot program envelope options for off-node mailboxes.

NOTE	This feature is available as a programmable option to subscribers, allowing them to change the selected option.

To disable an envelope option:

1. Select Voice Processor – Devices – Mailboxes – Local – *<**extension**>*.

2. Double-click **Envelope Settings**.

3. Double-click the message type **(E-Mail, Fax,** or **Voice**).

4. Clear the check box for the option that you want to disable.

E-MAIL SYNCHRONIZATION

This section contains the following information:

- **Feature Description** below

- Programming E-mail Synchronization on **page 973**

FEATURE DESCRIPTION

The Email Synchronization folder is for supporting the features included with Unified Voice Messaging (UVM) E-Mail Synchronization

Unified Voice Messaging (UVM) E-Mail Synchronization

Prior to v5.0, the "unified messaging" (UM) capability with Unified Voice Messaging (previously called Basic Voice Mail) was limited and did not allow true synchronization between the user's voice mailbox and e-mail client. With v5.0 or later, the UM capability for UVM has been enhanced to provide additional integration options and is now called E-Mail Synchronization.

This integration with an external e-mail server gives E-Mail Synchronization users the option to access, play, and manage their voice mail messages directly from their computer or Blackberry® e-mail client application. Each UVM voice mail message received by the user can also be delivered as an audio file attachment (.wav or .mp3) within an e-mail message sent to the user. For more information on setting up and using the E-Mail Synchronization feature on UVM, refer to the *UVM E-Mail Synchronization Administrator Guide* (document part no. 835.3286) and the *UVM E-Mail Synchronization User Guide* (document part no. 835.3287).

Note that this initial release of UVM E-Mail Synchronization does not provide a voice mail Web interface for accessing messages, nor does it provide any inbound or outbound faxing features.

E-mail Synchronization Levels

The E-mail Synchronization Level field specifies the level of integration. The options for this field vary depending on the Voice Processor type, as shown in **Table 162**.

The E-mail Synchronization level you select determines which additional fields in the E-mail Synchronization folder are programmable. See **page 973** for programming instructions and field descriptions.

NOTE	For Unified Messaging OSE v2.0, the values for the option previously programmed under the Fax/E-mail Forwarding folder in a v2.3 or earlier system remain the same and the fields automatically map to their new location under Voice Processor\Devices\Mailboxes\<*mailbox*>\E-mail Synchronization.
	Whenever you change the E-mail Synchronization Level for any mailbox to anything other than disabled, and if the E-mail Gateway is not set to SMTP or the E-mail server is not programmed, the following message appears: "Your change cannot be completed because E-mail Synchronization requires that the E-mail System be set to SMTP and the E-mail Server be programmed. Would you like to change your e-mail configuration?" Click **Yes** to go to the E-mail Gateway folder and make those programming changes, or click **No** to exit the dialog box without making changes.

Table 162:   E-mail Synchronization Levels for UVM

E-MAIL SYNCHRONIZATION LEVEL	DESCRIPTION	UVM VOICE PROCESSOR
Disabled	Delivers a voice message to the user's mailbox just as normal. No e-mail is sent.	✓
Forward Only	Delivers a voice message to the user's e-mail address each time they receive a new voice message in their voice mailbox. After the message is sent, the original message is deleted from their voice mailbox. This level *does not* provide integration between the user's e-mail client and voice mailbox.	✓
Forward and Copy	Delivers a copy of a voice message to the user's e-mail address each time they receive a new voice message in their voice mailbox. The original message can still be retrieved from their voice mailbox. This level *does not* provide integration.	✓
Enhanced Forward and Copy	Delivers new voice messages to the user's e-mail address and voice mailbox, and synchronizes those messages. For example, if the user deletes a voice message from their e-mail client, the message is also deleted in their voice mailbox. In addition, users receive a Welcome e-mail message and error notification e-mail messages if an error occurs. This level *does* provide integration between the user's e-mail client and voice mailbox.	✓
Basic Integration	Provides the user with direct access into the voice mailbox from the e-mail client. This level offers limited integration with the e-mail client for voice messages because a separate IMAP inbox is required for voice messages. The user has no configuration control over the format of the messages.	
Enhanced Integration[1]	Delivers new voice messages to the user's e-mail address and voice mailbox, and synchronizes those messages. For example, if the user deletes a voice message from their e-mail client, the message is also deleted in their voice mailbox. In addition, users receive a Welcome e-mail message and error notification e-mail messages if an error occurs. This level *does* provide integration between the user's e-mail client and voice mailbox.	✓
BlackBerry Enhanced Integration[1, 2]	Similar to the Enhanced Integration level, the BlackBerry Enhanced Integration level delivers new voice messages to the BlackBerry user's e-mail address and voice mailbox, and synchronizes those messages. This level *does* provide integration.	✓

1. To support the enhanced UVM E-Mail Synchronization levels (Enhanced Integration and Blackberry Enhanced Integration) for UVM E-mail Synchronization, the system require an appropriate system license. (Note that this is a single system-wide license and not a per-mailbox license.) This licensing also allows the MP3 file attachment option to be available to all of the E-Mail Synchronization levels. "If you are using Enhanced Integration or Blackberry Enhanced Integration for UVM E-mail Synchronization, you can program up to 100 (for a system without a PS-1) and 250 (for a system with a PS-1) mailboxes. If you attempt to program more than 250 mailboxes for E-mail Synchronization, a warning message appears and DB Programming cancels the changes."

2. If the BlackBerry Attachment Service (BAS) is hosted on a computer that uses Windows Server 2000 or 2008, the BAS may not support MP3 audio files on BlackBerry devices. Check with BlackBerry for supported servers and any other known issues regarding the BAS.

Supported E-mail Synchronization Fields

Table 163 indicates which fields in the E-mail Synchronization folder must be programmed for each E-mail Synchronization level. See **page 973** for field descriptions.

Table 163:   Programmable Fields for the E-mail Synchronization Folder

PROGRAMMABLE FIELDS	E-MAIL SYNCHRONIZATION LEVEL		
	DISABLED	FORWARD ONLY / FORWARD & COPY	ENHANCED INTEGRATION / BLACKBERRY ENHANCED INTEGRATION
Enable SSL for E-mail Server Connection			✓
E-mail Account Folder for Synchronization			✓
E-mail Server			✓
E-mail Account Password			✓
E-mail Account Username			✓
E-mail Address for Voice Messages		✓	✓
Allow User to Configure Settings			
Download Format for Mobile Web Page			
E-mail Client Message Format		✓	✓
Synchronize MWI with E-mail Client			✓
E-mail Address for Fax Delivery[1]			
Copy Fax to Sender			
Fax Delivery Destination[1]			
IMAP Synchronization Method[2]			
IMAP IDLE Timeout			
IMAP Polling Timeout			

1. This field is disabled if the voice processor type is UVM, which does not support faxing.

2. The option selected for the IMAP Synchronization Method determines which related fields are programmable:
   - *Event* (default): The IMAP IDLE Timeout field is enabled and the IMAP Polling Timeout field is disabled.
   - *Polling*: The IMAP IDLE Timeout field is disabled and the IMAP Polling Timeout field is enabled.

PROGRAMMING E-MAIL SYNCHRONIZATION

This section provides instructions for programming E-mail Synchronization.

NOTE	To support UVM E-mail Synchronization, you must have a UVM E-mail Synchronization software license uploaded to the system. To use the BlackBerry Enhanced Integration level, you also need a UVM Blackberry Integration software license.

To program E-mail Synchronization options:

1. Select Voice Processor – Devices – Mailboxes – *<mailbox>* – **E-mail Synchronization**.

2. Program the **E-mail Synchronization Level**. The E-mail Synchronization level specifies the level of integration for E-mail Synchronization. It is set to Disabled by default. Available options depend on the voice processor type. See Table 162 on **page 971** for option descriptions.

3. Program the following options (see Table 163 on **page 972** for supported fields for each E-mail Synchronization level):

   - **Enable SSL for E-mail Server Connection**: Indicates whether the integration connection is made using Secure Socket Layer (SSL). It is set to Yes by default. This field appears with a red "X" unless E-mail Synchronization Level is set to Enhanced Forward & Copy or Enhanced Integration.

   - **E-mail Account Folder for Synchronization**: Indicates the path to the account folder (such as Inbox, Voice Mail, and so forth) on the e-mail server to use for message synchronization. The folder name can be up to 127 characters. For example, to synchronize messages to the folder named "Voice Mail" in the user's e-mail client account, type Voice Mai**l** in the text box. To synchronize messages to the subfolder named "VM" under the "Inbox" main folder, type Inbox/VM in the text box. This field is set to Inbox by default. This field appears with a red "X" unless E-mail Synchronization Level is set to Enhanced Forward & Copy or Enhanced Integration.

   - **E-mail Server**: Indicates the name of the e-mail server for the account synchronization. The server name can be up to 127 characters. This field is blank by default. This field appears with a red "X" unless E-mail Synchronization Level is set to Enhanced Forward & Copy or Enhanced Integration.

   - **E-mail Account Password**: Indicates the password of the account to use for message synchronization. This is the user's network password. The password can be up to 40 characters. This field is blank by default. This field appears with a red "X" unless E-mail Synchronization Level is set to Enhanced Forward & Copy or Enhanced Integration.

   - **E-mail Account Username**: Indicates the user name of the account to use for message synchronization. This is the user's network username. The username can be up to 127 characters. This field is blank by default. This field appears with a red "X" unless E-mail Synchronization Level is set to Enhanced Forward & Copy or Enhanced Integration.

- **E-mail Address for Voice Messages**: Specifies the e-mail address to which the voice mail messages will be forwarded. This is also the e-mail address where welcome and error e-mails are sent for the Enhanced Forward and Copy and Enhanced Integration E-mail Synchronization levels. The content and format of the field depends on the e-mail system being used. The address can be up to 127 characters. If you enter an invalid character (: ; " \ | ( ) , < > '), an error tone occurs. For example, when using Lotus Notes, this field could be set to John Doe/Chandler/Mitel, and when using e-mail, it would be john_doe@mitel.com.

- **Allow User to Configure Settings**: Delegates account configuration control to the individual account users. It is set to Yes by default. This field appears with a red "X" unless E-mail Synchronization Level is set to Enhanced Forward & Copy or Enhanced Integration.

- **Download Format for Mobile Web Page**: Specifies which format to use when downloading voice messages from the mobile device Voice Mail Web page. Options include `.MP3` or `.wav` file formats.The default value is MP3. This field appears with a red "X" unless E-mail Synchronization Level is set to Enhanced Forward & Copy, Basic Integration, or Enhanced Integration.

NOTE	Most mobile devices do not support the `.wav` file format. Determine which format the user's mobile device supports and program this option accordingly.

- **E-mail Client Message Format**: Indicates whether the message that the user receives contains a WAV, MP3, or None. It is set to WAV by default. Setting this field to None turns the synchronization process into a simple message notification type of application. The synchronization process will be the same, but now in order to listen to the message, the user would need to call into the system. The e-mail message includes the callback information from the Message Notification Retrieval application.

NOTE	Unless the user's client application supports only MP3 files, Mitel recommends using the WAV file setting only. Refer to "Appendix E: Engineering Guidelines" in the latest *MiVoice Office 250 Installation Manual* for additional information.

- **Synchronize MWI with E-mail Client**: Synchronizes the Message Waiting Indication (MWI) support with the e-mail client. If set to Yes (default), the message lamp is turned off when the e-mail message is read. This field is applicable to the Enhanced Integration E-mail Synchronization Level only.

- **E-mail Address for Fax Delivery**: Specifies the e-mail address of the account that will receive incoming faxes. The fax is converted to a TIFF file and sent to the e-mail address as an attached file. The address can be up to 127 characters. If you enter an invalid character (: ; " \ | ( ) , < > '), an error tone occurs. This field appears with a red "X" if the Voice Processor type is UVM because UVM does not support faxing.

- **Copy Fax to Sender**: Enables the default operation for e-mailing a copy of a fax to a user's account as specified in the E-mail Address for Fax Delivery field (see above). The flag is enabled by default.

- **Fax Delivery Destination**: Specifies the extension of the fax machine that receives incoming faxes routed through this mailbox. This field appears with a red "X" if the Voice Processor type is UVM because UVM does not support faxing.

To program the Fax Delivery Destination extension number, use one of the following methods:

Method A

1. Select the current value and enter the new value in the box.
2. Press **ENTER**. A screen displays what is associated with the number you entered.
3. Click **OK**. The new number appears in the field.

Method B

1. Right-click on the existing extension number. An option box appears.
2. Click **Change Fax Delivery Destination**. A window appears prompting you to select the type of device to include.
3. Select the type of device and then click **Next**. The list of devices appears. (You can view them in a list only by selecting the List button.)
4. Highlight the extension number associated with the fax machine then click **Finish**. Your selection appears in Fax Delivery Destination field.

    • **IMAP Synchronization Method**: Determines how the IMAP server synchronizes messages between the server and an IMAP client account. By default, it is set to Event. This allows message changes on the remote IMAP server to be automatically recognized by the synchronization client, as they occur. If the IMAP server does not support the IMAP IDLE command, select **Polling**. This forces the synchronization client to query for changes on the account on the remote server. If you do not know if the remote IMAP server supports the IDLE command, default to Event. At run time the synchronization client automatically detects if the IDLE command is supported. If not supported, the client uses the Polling method. Polling can be used as default to reduce load on the unit. This field appears with a red "X" unless the E-mail Synchronization Level is set to Enhanced Forward & Copy or Enhanced Integration.

    • **IMAP IDLE Timeout**: Determines the maximum length that the IDLE command will wait for the IMAP server. The field appears with a red "X" if the Integration Synchronization Method is set to "Polling." The range is 1–30 minutes; the default value is 20 minutes. This field appears with a red "X" if the IMAP Synchronization Method is set to "Polling" as it only applies for Event or unless E-mail Synchronization Level is set to Enhanced Forward & Copy or Enhanced Integration.

    • **IMAP Polling Timeout**: Indicates how often the synchronization process polls for message updates. Use this when the server does not support the IDLE command. The range is 1–20 minutes; the default value is 10 minutes. The field appears with a red "X" if the IMAP Synchronization Method is set to "Polling" as it only applies for Event or unless E-mail Synchronization Level is set to Enhanced Forward & Copy or Enhanced Integration.

5. Click out of the field or press **Enter** to save the change.

To determine the language used for mailbox displays, such as E-Mail Synchronization:

See **page 994** for details.

In order for E-Mail Synchronization to work properly, the following system settings must also be properly configured in DB Programming:

- E-Mail Gateway (E-Mail System must be set to SMTP)

- IP Settings (Base Server Hostname and Domain Name must be configured)

To configure the E-Mail Gateway folder:

1. Select System – **E-Mail Gateway**.

2. Change the **E-mail System** setting to **SMTP**.

3. Configure all of the available fields within the E-Mail Gateway folder (see "E-Mail Gateway" on **page 942**). For details about the fields, refer to the *MiVoice Office 250 DB Programming Help*. Note that the E-Mail SMTP Server field must use a fully-qualified domain name (FQDN) and not an IP address.

To configure the IP Settings folder:

1. Select System – **IP Settings**.

2. Make sure that the **Base Server Hostname** and **Domain Name** fields are properly configured. See "General IP Settings" on **page 603** for details.

RECORDING LENGTH

You can view the number of seconds used by the mailbox directory name and the primary and alternate greetings. The information is shown for reference only. It is *not* programmable.

To view the Recording Lengths:

1. Select Voice Processor – Devices – Mailboxes – Local – *<extension>*.

2. Double-click **Recording Lengths**.

MESSAGE LIMITS

The message limits that can be programmed for each mailbox include the following:

- **Maximum Mailbox Message Capacity**: The mailbox can be programmed to hold up to 600 minutes of messages or have unlimited message capacity. The range is 0–600 minutes; the default value is 30 minutes. Enter 0 for unlimited capacity.

- **Maximum Non-Subscriber Message Length**: Maximum non-subscriber messages can be set to a value between one minute and the Maximum Mailbox Message Capacity setting. The range is 1 minute to the Maximum Mailbox Message Capacity value; the default value is 5 minutes.

- **Maximum Outgoing Message Length**: The length of outgoing messages by this subscriber. The range is 1–120 minutes; the default value is 5 minutes.

To change a message limit:

1. Select Voice Processor – Devices – Mailboxes – Local – <***extension***>.

2. Double-click **Message Limits**.

3. Select the Message Limit.

4. In the **Value** column, type the new number in the box. *If you enter an invalid number,* a warning message appears.

5. Click out of the field or press **ENTER** to save the change.

SUBSCRIBER STATISTICS

You can view subscriber usage statistics. The statistics reflect the period since the last date that the statistics were cleared. Statistics are cumulative and remain as such until cleared using Report Parameters. See "Report Parameters" on **page 1025**. The information is shown for reference only. It is *not* programmable. Statistics are as follows:

*   **Last Logon Date (and Time)**: Reflects the most recent date and time of the last valid logon by the subscriber. (If the System Administrator makes any change to a subscriber's personal options from the System Administrator's mailbox, no change will be made to this field.)

*   **Number of New Messages**: Reflects a count of the number of messages in a subscriber's new message queue. It is the same number that is reported to the subscriber when he or she logs in to the mailbox.

*   **Number of Saved Messages**: Reflects a count of the number of messages stored in the saved message queue for the mailbox. It is the same number that is reported to the subscriber when he or she logs in to the mailbox.

*   **Mailbox Percent Full**: Shows the actual percentage of maximum mailbox message capacity used.

*   **Number of Times Mailbox Was 80% Full**: Shows the number of times the mailbox reached 80% of its maximum message capacity.

*   **Number of Times Mailbox Was Full**: Displays the number of times a mailbox reached its maximum message capacity.

*   **Number of Messages Sent**: Reflects a count of the number of times a subscriber records and sends a message (to one mailbox or a group list of mailboxes), replies to a message sent by another subscriber, or forwards a message with comments.

*   **Number of Messages Received**: Shows the number of messages a subscriber has received regardless of where the messages came from (subscriber, non-subscriber, or system).

*   **Total Length of New and Saved Messages**: Reflects a combined total of the amount of time represented by the "Number of New Messages" and "Number of Saved Messages" fields.

*   **Number of Times 3 Bad Passcodes Were Entered**: Increases each time a single call includes three attempts to enter a mailbox and the caller uses an incorrect mailbox/password combination.

To view the subscriber statistics:

1. Select Voice Processor – Devices – Mailboxes – Local – *<**extension**>*.

2. Double-click **Statistics**.

DIAL-0 DESTINATIONS

The dial-0 destination is where calls are sent if callers dial 0 for the operator. Dial-0 destinations are used for Voice Mail and Automated Attendant applications.

PROGRAMMING THE DAY AND NIGHT DIAL-0 DESTINATION TYPES

The following options select the type of device that is used for the day and night operator destinations:

*   None: Operator access is denied from Voice Mail and Auto Attendant applications.

*   **Extension**: The system automatically transfers the call to any system phone application (including a STAR application) or hunt group extension. The system software supports STAR applications as a valid Dial-0 Destination.

*   **Mailbox**: The system automatically transfers the caller to the mailbox specified in the Dial-0 Destination (see the following section) when the caller dials 0.

*   **Operator**: If the operator destination is set to **Operator**, the caller is transferred to the attendant.

To program the Day or Night Dial-0 Destination type:

1. Select Voice Processor – Devices – Mailboxes – Local – *<**extension**>*.

2. Select **Dial-0 Day** (or **Night**) **Destination Type**.

3. In the **Value** column, select the option from the list (**None**, **Extension**, **Mailbox**, or **Operator**).

4. Click out of the field or press **ENTER** to save the change.

PROGRAMMING THE DIAL-0 DESTINATION

If you select day or night destination types, as described in the previous section, you can select the specific device that serves as the operator destination.

To select the Dial-0 Destination:

1. Select Voice Processor – Devices – Mailboxes – Local – *<**extension**>*.

2. Select **Dial-0 Day** (or **Night**) **Destination Type**.

3. In the Dial-0 Destination column, right-click the existing value, and then click **Change Dial-0 Destination**. A window appears prompting for the device type to include.

4. Select the device types (you can use the SHIFT or CTRL key to select more than one item), and then click **Next**. The items with details appear. To view items in a list only, click **List**.

5. Select the desired destination, and then click **Finish**. The selection appears in the list.

REMOTE MESSAGING

UVM systems only. Remote Messaging is a subscriber feature that is enabled through database programming. Subscribers may program a series of specific telephone numbers (a "cascade") for the Voice Mail system to call when new messages are received by their mailboxes. See **page 986** for a sample notification cascade setup. For end-user instructions, refer to the *Unified Voice Messaging and Embedded Voice Mail Card User Guide*, part number 835.3205.

Remote Messaging places a call to subscribers when their mailboxes receive new voice messages. Using "cascade levels" of up to nine phone numbers (see **page 980**), the voice mail system calls each number until it successfully connects to a device (for example, a home phone, a mobile phone, or a pager). The figure below shows an example of Remote Messaging routing.

NOTE	You cannot enable Remote Messaging unless you have programmed cascade levels (see **page 981**). If you try to enable Remote Messaging without any cascade levels, an error is displayed.

Voice mail records the message.

Depending on when the message was received, voice mail calls either your primary or alternate cascade.

The cascade level 1 number (for example, your home phone) receives the call from voice mail.

There is no answer, so voice mail places another call to the cascade level 2 number.

The cascade level 2 number (for example, your mobile phone) receives the call from voice mail, and you answer the call.

You are notified of the new message ("You have a voice message from...").

To enable Remote Messaging:

1. Select Voice Processor – Devices – Mailboxes – Local – <***extension***> – **Remote Messaging**.

2. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

3. Click out of the field or press **ENTER** to save the change.

PRIMARY AND ALTERNATE MESSAGE NOTIFICATION

Each mailbox can have a Primary and/or an Alternate notification cascade of numbers. The notification numbers are programmed to be accessed during specific days and times.

The Voice Mail system monitors subscriber mailboxes continuously. During monitoring, if new messages have been received, the system determines if Primary Notification is turned on. If it is, the system checks the day and time programming. If the current time is within the programmed notification day and time, the Voice Mail system makes the notification call(s) to the numbers in the Primary Notification cascade. If Primary Notification is turned off or if the day or time does not match, the Voice Mail system immediately makes the same check for Alternate Notification and possibly place calls to the numbers in the Alternate Notification cascade.

When Voice Processing makes a notification call, it moves through the appropriate notification cascade level by level as determined by the notification and retry programming described in the following sections. When the system reaches the last level in the cascade, it returns to level 1 and begin again, if necessary.

If the system is not able to place the call within the parameters of the programmed notification day and time (due to busy facilities or invalid programming), the mailbox receives a message stating that notification could not be completed (the mailbox owner hears the message next time he accesses the mailbox.).

The number of calls Voice Processing can process simultaneously is determined by the Maximum Number of Outgoing Calls flag. It defaults to 2, but can be programmed to use all of the enabled Voice Processing voice channels.

NOTE	The number of calls Voice Processing can process simultaneously is also limited by the available time slots (maximum channel allocations) for the application. For more information about maximum channel allocations, refer to the "Specifications" chapter in the *MiVoice Office 250 Installation Manual* . An outgoing message notification call that exceeds the number of channels allocated to the Message Notification application results in an error message.

If you enable Remote Messaging (see **page 979**), you must program either **Primary** or **Alternate Message Notification** parameters.

NOTE	If Remote Messaging is enabled, this feature is available as a programmable option to subscribers, allowing them to change the selected option.

To view or edit Primary or Alternate Message Notification Options:

Select Voice Processor – Devices – Mailboxes – Local – <***extension***> – **Primary** (or **Alternate**) **Message Notification**. The following options are shown in the right pane:

- **Cascade Levels:** Double-click **Cascade Levels** to view the list of cascades. The list appears blank if no cascade levels have been programmed. For cascade programming options, see **page 982**.

NOTES	Primary and alternate cascade levels are not equipped in pairs. For example, if you add or delete a primary cascade level, a corresponding alternate cascade level is ***not*** automatically added or deleted.
	When a Remote Message Notification call is placed, the prompt "You have received a remote message . . ." starts to play as soon as a programmable timer expires (default value is 3 seconds). Voice Processing plays the prompt five times before disconnecting the call. Playing out these prompts five times covers the (programmable) maximum number of rings and still plays at least one iteration of the prompt. User's must press pound (#) or star (*) to stop the prompts. The programmable delay only extends the prompt replays. Users will probably hear the end of a previous prompt, but they can hear the entire prompt if they want to.

- **Notification Category:** You can set message notification to place the notification call for all messages or only when priority messages are received. The default state is all messages.

    a. Select option from the list.

    b. Press **ENTER** or click another field to save the change.

- **Call For Each New Message**: This determines whether the Voice Processor should attempt message notification *every* time a message is received (that meets the notification category set above), or only when a message is received and no other messages are waiting to be picked up.

    To change the setting:

    a. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

    b. Press **ENTER** or click another field to save the change.

- **Start/Stop Time:** Message notification can be enabled for any time period, up to 24 hours per day. The default value is 8:00AM to 5:00PM. If start and stop times are the same, notification will be enabled 24 hours per day.

    To set the time:

    a. Select the current Value, then select the hours, minutes, or AM/PM field, and use the arrows to scroll to the desired setting.

    b. Select another field or press **ENTER** to save the change.

- **Days Of The Week:** Message notification can be set to place notification calls only on certain days. (This defaults to Monday through Friday.)

    To enable a day:

    a. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

    b. Press **ENTER** or click another field to save the change.

ADDING CASCADE LEVELS

To add cascade levels:

1. Select Voice Processor – Devices – Mailboxes – Local – *<extension>* – **Primary** (or **Alternate**) **Message Notification**.

2. Right-click in the right pane, and then select **Add cascade level(s)**.

3. Select the starting ID and the number of levels you want to add. (For example, to add 9 levels with IDs 1–9, select 1 as the **Starting ID** and 9 as the **Number of IDs**.)

4. Click **OK**. The levels are added to the list with default values. You can then change the Notification Number for any cascade by entering the new number in its text box. You can also set the Notification flag by placing a check in the check box in this screen or in the cascade settings described below.

DELETING CASCADE LEVELS

To delete cascade levels:

1. Select Voice Processor – Devices – Mailboxes – Local – *<extension>* – **Primary** (or **Alternate**) **Message Notification**.

2. Right-click the cascade level that you want to remove, and then select **Remove Selected Item**.

PROGRAMMING CASCADE LEVEL OPTIONS

To program the cascade level options:

1. Select Voice Processor – Devices – Mailboxes – Local – *<extension>* – **Primary** (or **Alternate**) **Message Notification**.

2. Double-click **Cascade Levels**.

3. Double-click the cascade level you want to program to view the following options:

- **Timers and Limits:** Double-click Timers and Limits to program the following. If necessary, enter a new Value, then press **ENTER** or click another field to save the change.

  - *Number Of Call Attempts*: Determines how many times the Voice Processor will attempt to complete a call to this cascade level before moving to the next level. The range is 1–1000 call attempts; the default value is 1.

  - *Number Called Busy Timer*: Defines the amount of time the Voice Processor will wait between outgoing call attempts whenever a busy signal is encountered during a remote message notification attempt to a personal number or pager. The range is 0–1440 minutes; the default value is 5.

  - *Personal Number No Answer Timer:* Defines the amount of time the Voice Processor will wait between outgoing call attempts whenever a message notification attempt to a personal number is unanswered. The range is 0–1440 minutes; the default value is 30.

  - *Pager Notification Retry Timer*: Reflects the amount of time the Voice Processor will wait between outgoing call attempts when the notification number is a pager. The range is 0–1440 minutes; the default value is 20.

- **Enable Notification**: Each cascade can be enabled or disabled individually. To change the setting, click the check box to place a mark in it and set the flag to **Yes**. (Or click an existing mark to disable it and change the flag to **No**.) Press **ENTER** or click another field to save the change.

- **Call Progress Detection:** This determines if the Voice Processor will perform call progress detection after dialing the notification number and outgoing termination fields on outside calls. If it is enabled, the system will analyze call progress tones after sending the number. If it is disabled, the system will assume the call has been answered, without analyzing tones. This should only be disabled in cases where call progress tones are not used (lighting a message lamp) or when the tones used by the destination cannot be recognized by the system (some pager destinations). To change the setting, select the check box to remove the mark in it and set the flag to **No**. (Or click it again to enable it and change the flag to **Yes**.) Press **ENTER** or click another field to save the change.

- **Notification Type**: The message notification number can be identified as a personal number (a person will answer the call) or a pager. (This defaults to personal number.) Select the option from the list. Press **ENTER** or click another field to save the change.

- **Notification Destination Type:** This indicates whether the notification call is going to an intercom (IC) or outside (CO) destination. This can also be programmed by the mailbox Subscriber. Select the drop-down list box and scroll to the desired setting. Press **ENTER** or click another field to save the change.

- **Notification Destination:** This is the number (outside number, local extension, or off-node extension) to be notified when the mailbox receives a message. If the Notification Destination Type is IC, right-click the field and select **Change Notification Destination**. Program the destination device. If the Notification Destination Type is CO, enter the outside number in the text box. Press **ENTER** or click another field to save the change. The outside number can contain up to 24 characters including digits (0–9, #, and *) or P for pauses. This number should not include pager display numbers; they are programmed below.

- **Pager Dial String:** This digit string can contain up to 54 characters. It should include any digits that the paging company requires when the call is answered, the pager LCD number, and the pager termination code, if needed. Valid entries include: any digit 0–9, #, *, P for pause. Also, if you want to have the pager show the number of the mailbox that placed the call, you can use an M in the dial string. To show the number of waiting messages for the mailbox, use an N. For example, if the Pager String is programmed as 9619000*MN# and a pager call is placed by mailbox number 1234 which has 3 waiting messages, the pager display would show 9619000*12343 (the # is used as a call termination digit). Enter the dial string in the text box, then press **ENTER** or click another field to save the change.

- **Outgoing Access:** This is used to identify the trunk group that will be used for placing remote notification calls. To change the trunk group, right-click **Value** and select **Change Outgoing Access**. In the first window, scroll to CO Trunk Groups, then click **Next**. In the next window, select the desired trunk group, then click **Finish** to exit.

- **Outgoing Access Prefix:** This is the dial string that the system uses before an outgoing number, if any. Enter the dial string in the text box and then press **ENTER** or click another field to save the change.

- **Outgoing Termination:** This is the dial string that the system uses to terminate an outgoing number, if any. Enter the dial string in the text box, then press **ENTER** or click another field to save the change.

ALL MESSAGES, PRIORITY ONLY, AND EACH NEW MESSAGE FLAGS

When the mailbox is being programmed for remote notification, message notification can be set to place the notification call when any message is received or only when priority messages are received. (This defaults to "All Messages.") Another mailbox flag, called "Call For Each New Message," determines when the voice processor attempts to make the notification call when more than one message is waiting to be heard:

- If the Call For Each New Message flag is *enabled,* the voice processor attempts *immediate* notification each time a new message (or priority message, depending on the option selected) is received, regardless of the status of the applicable retry timer. If a retry is in progress, it places the call to the number on the cascade table that would have received the retry call, as described below. (For example, if the voice processor was on level 3 of a cascade, attempting retries, it does not start over again at level 1. It places the call to level 3.)

- If the Call For Each New Message flag is *disabled,* new messages do not generate a remote notification attempt until the subscriber has logged on and listened to the waiting message(s). (However, retry calls are attempted if the applicable retry timer expires, as described "Retry Call Attempts" on **page 984**.)

RETRY CALL ATTEMPTS

The mailbox can be programmed to retry notification calls when it encounters a busy trunk or trunk group. Each cascade level can be programmed with a Number of Call Attempts value of 1-1000 calls. If the mailbox user accesses the mailbox between the time the message is received and when remote notification is successful, the system stops attempting remote notification. It is assumed that the mailbox user listened to the message when the mailbox was accessed. However, the **MSG** button at the phone remains lit if there are any messages that have not been heard.

Each level also has a Number Called Busy timer, and a Pager Notification Retry timer or Personal Number No Answer timer. These timers determine how long the voice processor waits before making the next notification attempt when messages are waiting to be heard (unless it is overridden by the Each New Message flag). These timers can be set to a value of 0–255 minutes. If set to 0, it retries the cascade level immediately, for as many attempts as allowed by the Number of Call Attempts field. If the message(s) have not been picked up by then, the system moves to the next cascade level. The default setting for the Pager Notification Retry timer is 20 minutes. The Number Called Busy and Personal Number No Answer timers default to 5 minutes.

> ### NOTICE
> The 0-minute timer value should be used cautiously. If the system is forced to make several calls in quick succession, it impairs Voice Processing ability to place other outgoing calls. It should be used primarily with cascades in which all of the members wish to receive notification at approximately the same time. In this case, all levels except the last could be set to 0-minute retries with 1 allowed call attempt. However, the last level in the cascade should be programmed with a longer Pager Notification Retry timer so that the system does not continuously cycle through the cascade levels. (Continuous cycling is also prevented somewhat by a per-mailbox 15-call limit. Due to FCC regulations, Voice Processing can allow a mailbox to make only 15 calls in rapid succession without a 10-minute pause.)

NOTIFICATION NUMBERS

Each cascade level has a programmed notification number that can be an extension number or an outside telephone number. When an outside telephone number is used, the system accesses an outgoing trunk using the code programmed for that level that contains a pre-programmed trunk access code and followed by an outgoing access termination feature code, if needed.

Notification calls to outside telephone numbers can be sent to a pager or to a number where a person answers (personal number).

- If the call goes to a pager, the system dials the programmed Pager Dial String, then disconnect the call. The dial string should include any digits that the paging company requires when the call is answered, the pager LCD number, and the pager termination code, if needed. Also, if desired, the pager string can be programmed to show the number of the mailbox that placed the call and/or the number of waiting messages in the mailbox.

- If the call goes to a personal number, the system plays a prompt announcing that the call is from Voice Mail, play the directory name or mailbox number that originated the call, and play user instructions. The listener can then enter the mailbox number password to gain access to the mailbox and hear the message. The listener has three chances to enter a correct password. If the system receives an incorrect password three times, the Voice Mail system disconnects the call immediately, and the attempt is considered unsuccessful.

SAMPLE PAGER PROGRAMMING

The following is a sample programming sequence for enabling pager notification. In the example, the pager number is 555-2500, trunk group 1 is used for notification, the display should show 961-9000 when a message is left in mailbox 1001, and the paging company requires a # after the digits.

1. In Voice Processor Programming, create a Message Notification/Retrieval application.

2. If desired, set toll restriction for the Message Notification/Retrieval application in Voice Processor\Devices\Applications\Class of Services.

3. Ensure the Message Notification/Retrieval application has outgoing access for trunk group 92001.

4. In Mailbox Programming for mailbox 1001, enable **Remote Notification** and program the following for the primary and/or alternate message notification:

   a. **Notification Category**: Determine whether all messages or only priority messages cause message notification.

   b. **Call For Each New Message flag**: Determine if you want to call the pager every time a new message arrives, even while there is an unheard message waiting.

   c. **Start Time, Stop Time, and Day of Week**: Set the parameters for when you want the pages to be sent.

   d. **Cascade Table**: Program the cascade table(s) needed. In the example, the table would have:

      - Notification Type: **Pager**
      - Notification: **Enabled**

- Notification Destination Type: **CO**

- Notification Number: **555-2500**

- Outgoing Access: **92001** (Trunk Group 1)

- Pager Dial String: **961-9000#**

- Timers: Can be adjusted to suit the pager user's requirements.

SAMPLE NOTIFICATION CASCADE

Step **4d** on **page 985** shows how each level in a Primary and Alternate cascade is programmed to meet a customer's needs. In the example, the pager dial string (9619000*MN#) includes the Voice Processing telephone number (961-9000), an asterisk to separate digits, the mailbox number (M), the number of messages in the mailbox (N), and a pager termination digit (#).

Table 164:   Example Notification Cascade Entries

CUSTOMER WANTS:	CASCADE LEVEL:	PROGRAM THESE PARAMETERS:
During weekly business hours (8–5), customer wants to be notified for all messages, each time a message is received...	Primary Notification	Notification Category: All Messages Call For Each New Message: Enabled Start Time/Stop Time: 8:00 AM/5:00 PM Days of Week: Monday-Friday
First, call customer's desk (extension 1234) for up to 3 attempts, 1 minute apart...	Primary – Level 1	Enable Notification: Yes Notification Destination Type: IC Notification Number: 1234 Notification Type: Personal Number of Call Attempts: 3 calls Number Called Busy: 1 minute Personal Number No Answer: 1 minute
If customer does not answer at desk, customer wants Voice Mail to call car phone (555–1000) up to 5 times, every three minutes...	Primary – Level 2	Enable Notification: Yes Notification Destination Type: CO Notification Number: 5551000 Notification Type: Personal Outgoing Access: 92001 (trunk group 1) Outgoing Access Termination: # Number of Call Attempts: 5 calls Number Called Busy: 3 minutes Personal Number No Answer: 3 minutes
If customer does not answer at car phone, customer wants pager (555–2000) to be called...	Primary – Level 3	Enable Notification: Yes Notification Destination Type: CO Notification Number: 5552000 Notification Type: Pager Pager Dial String: 9619000*MN# Outgoing Access: 92001 Outgoing Access Termination: # Number of Call Attempts: 1 call Number Called Busy: 0 minutes Pager Notification Retry: 0 minutes

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Table 164: Example Notification Cascade Entries (continued)

CUSTOMER WANTS:	CASCADE LEVEL:	PROGRAM THESE PARAMETERS:
Immediately after calling customer pager, customer wants it to call assistant's pager (555–3000) right away.	Primary – Level 4	Enable Notification: Yes Notification Destination Type: CO Notification Number: 5553000 Notification Type: Pager Pager Dial String: 9619000*MN# Outgoing Access: 92001 Outgoing Access Termination: # Number of Call Attempts: 1 call Number Called Busy: 255 minutes Pager Notification Retry: 255 minutes
After business hours and during the week, customer wants to be notified for only priority messages, each time a priority message is received...	Alternate Notification	Notification Category: Priority Only Call For Each New Message: Enabled Start Time/Stop Time: 5:00 PM/8:00 AM Days of Week: Monday-Friday
Customer wants pager called (555–2000) for all priority messages.	Alternate – Level 1	Enable Notification: Yes Notification Destination Type: CO Notification Number: 5552000 Notification Type: Pager Pager Dial String: 9619000*MN# Outgoing Access: 92001 Outgoing Access Termination: # Number of Call Attempts: 1 call Number Called Busy: 0 minutes Pager Notification Retry: 0 minutes

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MAILBOX INITIALIZED

This field indicates whether the subscriber has initialized the mailbox and recorded a name in the company directory. The field is shown for reference only and is not programmable.

To view the Mailbox Initialized field:

Select Voice Processor – Devices – Mailboxes – Local – <***extension***>. The Mailbox Initialized option is shown in the right pane.

RECEIVE ONLY

You can program mailboxes as "Receive Only." Receive Only mailboxes cannot be used to send messages. By default, mailboxes do not have this option enabled.

To make the mailbox receive-only:

1. Select Voice Processor – Devices – Mailboxes – Local – <*extension*> – **Receive Only**.

2. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

3. Click out of the field or press **ENTER** to save the change.

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ALLOW TRANSFER METHOD PROGRAMMING

This option determines whether the subscriber (or voice mail administrator) can change the Transfer Method using the voice mail Personal Options prompts. If enabled, it allows user programming. If disabled, you must change the Transfer Method in DB Programming.

1. Select Voice Processor – Devices – Mailboxes – *<node>* – *<extension>* – **Allow Transfer Method Programming**.

2. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

3. Click out of the field or press **ENTER** to save the change.

PLAY RECORDING INSTRUCTIONS

If this option is enabled, the system prompt that tells the caller to leave a message after the beep plays after the primary or alternate greeting. If disabled, the beep occurs as soon as the primary or alternate greeting ends, and the system prompt does not play. By default, this option is enabled.

Use of this option is subject to the following conditions:

- If the system greeting is selected, the instructions cannot be disabled.

- If the primary greeting is selected and has not been recorded, the instructions cannot be disabled.

- If the secondary greeting is selected and has not been recorded, the instructions cannot be disabled.

To disable the instructions that play after the primary or alternate greeting:

1. Select Voice Processor – Devices – Mailboxes – Local – *<extension>* – **Play Recording Instructions**.

2. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

3. Click out of the field or press **ENTER** to save the change.

AUTO ATTENDANT TRANSFER PROMPT

The Auto Attendant Transfer Prompts determines whether the transfer prompt ("Please hold while your call is being transferred to...") plays after a caller has entered the extension number of the phone associated with this mailbox. This applies to calls transferred by Automated Attendant and Call Routing Announcement applications, including transfers to the operator's mailbox or extension ID. This option is enabled by default.

To disable the prompt:

1. Select Voice Processor – Devices – Mailboxes – *<node>* – *<extension>* – **Auto Attendant Transfer Prompt**.

2. In the **Value** column, clear the check box. The field changes to **No**. To enable the option, select the check box.

3. Click out of the field or press **ENTER** to save the change.

DELIVER HANGUP MESSAGE (WHEN ANI IS AVAILABLE)

This option determines when to leave an "indication message" if the caller hangs up. This indication message is a system-generated message that states there was an incoming call to the voice mailbox but that no voice message was left. The purpose of this indication message is to capture Caller ID (CLID) information for the call and display it on the phone. If the Deliver Hangup Message (when ANI is available) field is set to Yes (default), the system leaves indication messages for calls that are shorter than the Shortest Message Allowed timer (Voice Processing\Timers and Limits\Timers and Limits). If set to No, the feature is disabled, and the system does not leave an indication message.

To disable the Deliver Hangup Message (when ANI is available) option:

1. Select Voice Processor – Devices – Mailboxes – Local – *<extension>* – **Deliver Hangup Message (when ANI is available)**.

2. In the **Value** column, clear the check box. The field changes to **No**.

3. Click out of the field or press **ENTER** to save the change.

SWAP "7 FOR SAVE" AND "9 FOR DELETE" MESSAGE KEYS

NOTES	This feature is available as a programmable option to subscribers, allowing them to change the selected option.

Swap "7 for Save" and "9 for Delete" Message Keys feature applies to UVM and only.

You can use the Swap "7 for Save" and "9 for Delete" Message Keys to reverse the 7 and 9 digit operations in subscribers' Message Options. By default, the 7 digit is used for SAVE and the 9 digit is used for DELETE. With the swap feature, you can use the 7 digit for DELETE and the 9 digit for SAVE, just like cell phone voice mail system functionality.

This helps users who are familiar with cell phone functionality from accidentally deleting messages by pressing 9, intending to save messages. Subscribers can program this feature for their mailboxes (see below) or you can program this feature for all subscribers in the voice processing system settings.

To swap the keys for individual mailboxes:

1. Select Voice Processor – Devices – Mailboxes – Local – *<extension>* – **Swap "7 for Save" and "9 for Delete" Message Keys**.

2. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

3. Click out of the field or press **ENTER** to save the change.

DESIGNATE THIS MAILBOX FOR PLAY ONLY

Applies to UVM non-associated mailboxes only. The UVM "play-only" mailbox option prevents the caller from leaving a message. The "play-only" mailbox is similar to an "announcement-only" application where it plays the designated greeting once, and then the call automatically disconnects after the greeting finishes playing.

This feature allows a subscriber to record a custom message without requiring administrative access to voice mail. For example, a teacher (the "subscriber") could record a homework assignment message for students who missed class that day. The students could then retrieve the message from the mailbox, but they would not be able to leave messages in the mailbox.

Each UVM mailbox in DB Programming, has an option to indicate if it is "play-only." The default value is set to "No" for all mailbox types.

To enable or disable an unassociated mailbox for play only:

1. Select Voice Processor – Devices – Mailboxes – Local – *<extension>* – **Designate This Mailbox for Play-Only**.

2. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

3. Click out of the field or press **ENTER** to save the change.

PASSCODE

You can program the passcodes that subscribers use to access their mailboxes.

NOTES	To provide system security, *all* mailboxes should have a passcode. To make the passcodes difficult to guess, they should **not** match the mailbox number or consist of one digit repeated several times. The default passcode should be changed the first time the user logs in. This is especially important in the voice mail administrator mailbox, which allows programming access to the voice processor.
	This feature is available as a programmable option to subscribers, allowing them to change the selected option.

To program a mailbox passcode:

1. Select Voice Processor – Devices – Mailboxes – *<node>* – *<**extension**>*.

2. Right-click **Passcode**, and then select **Edit Passcode**. The Edit Passcode dialog box appears.

3. In the **New Passcode** box, type the passcode (up to 12 digits, using digits 0–9). The digits appear as asterisks (***) in the box.

4. In the **Confirm passcode** box, retype the passcode.

5. Click **OK** to exit and save the passcode.

GREETING

You can enable one of the following greetings:

- **System**: The default greeting that callers hear when they access subscribers' mailboxes. The following is an example system greeting. "<*Subscriber's name*> is not available. After the tone please record your message. When finished leaving your message, hang up, or press # for more options."

- **Primary**: The "standard" greeting used by subscribers who are unavailable to take calls. Subscribers record their own primary greetings.

- **Alternate**: An alternate greeting that subscribers can use for vacations, days off, and so forth.

> **NOTE** This feature is available as a programmable option to subscribers, allowing them to change the selected option.

To select the voice mail greeting:

1. Select Voice Processor – Devices – Mailboxes – Local – <*extension*> –**Greeting**.

2. In the **Value** column, select the greeting from the list (Primary, Alternate, or System Greeting). Even if you select Primary or Alternate greeting, the System greeting plays until the subscriber records the selected greeting.

3. Click out of the field or press **ENTER** to save the change.

TRANSFER METHOD

(Applies to associated mailboxes only.) When a call is received by an automated attendant and the caller enters an extension number, the Transfer Method determines how the call is transferred. Transfer Methods are as follows:

- **Announce-Only:** The caller is asked to state their name, and then the call is transferred to the associated extension number. When the phone user answers the transfer, the system plays the caller's name and completes the transfer.

- **Screened:** The caller is asked to state their name, and then the call is transferred to the associated extension number. When the phone user answers the transfer, the system plays the caller's name. The phone user has the options of replaying the name, sending the call to voice mail (if the extension has a mailbox), transferring the call to another extension, accepting the call, or rejecting the call.

- **Unannounced:** The call is transferred to the associated extension number after the system checks the phone to determine its status (busy, available, ringing, and so forth).

> **NOTE** This feature is available as a programmable option to subscribers, allowing them to change the selected option.

To select a Transfer Method:

1. Select Voice Processor – Devices – Mailboxes – <*node*> – <*extension*> –**Transfer Method**.

2. In the **Value** column, select the option from the list.

3. Click out of the field or press **ENTER** to save the change.

MESSAGE NOTIFICATION PHONE

UVM systems only. Each mailbox has a phone that receives message notification whenever the mailbox receives a message. This is usually the same number as the mailbox (associated mailbox). However, a different number can be used for the mailbox message notification phone (non-associated mailbox). When a mailbox is shared by several phones or belongs to a hunt group, a specific phone must be designated to receive the message notification. If non-associated mailboxes are programmed, you must disable the Validate Voice Mailbox Number feature (see **page 755**) to allow subscribers to dial the non-associated mailbox number.

- **Associated**: When the extension number, message notification number, and mailbox number for a phone all match, that phone user can forward calls to Voice Mail and have the call go directly to the associated mailbox personal greeting (even if other mailboxes use that extension number as their message notification extension).

- **Non-Associated**: The extension number assigned as the Message Notification Phone for a non-associated phone is an extension number that does not match the mailbox number. (For example, a hunt group pilot number can have a mailbox, but the message notification must be sent to a specific phone so that a message lamp can be lit or message notification signal can be sent.) If a phone user's extension number does not match a mailbox number, and the phone user forwards calls to Voice Mail, the caller hears the main Voice Mail greeting and must enter a mailbox number.

NOTE	If there are non-associated mailboxes in the Voice Processing system, the system database must have the Validate Voice Mailbox flag disabled. When enabled, this flag prevents users from dialing mailbox numbers that do not match valid extension numbers.

When a mailbox receives a message, the message notification phone is signaled. If it is a phone, the **MSG** button is lit, and the display shows that a message has been received. A single-line set receives message waiting signals, if enabled system-wide.

To designate the Message Notification phone:

Use one of the following methods:

Method A

- **a.** Select Voice Processor – Devices – Mailboxes – Local – *<non-associated mailbox extension>* –**Message Notification Phone**.

- **b.** In the **Value** column, select the current value, and then enter the new extension number in the box.

- **c.** Click out of the field or press **ENTER** to save the change.

Method B

- **a.** Right-click the existing extension number, and then select **Message Notification Phone**. A dialog box appears prompting for the device type to include.

- **b.** Select the device type, and then click **Next**. The list of devices with details appears. To view the items in a list only, click **List**.

- **c.** Select the appropriate phone, then click **Finish**. The selection appears in the Message Notification Phone field.

TIME ZONE

If the mailbox is located in a different time zone than the external voice processing system, you can set the Time Zone option to match voice processing system location. This allows the time stamp on voice mail messages to reflect the correct time for that mailbox location.

To set the time zone:

1. Select Voice Processor – Devices – Mailboxes – Local – *<extension>* –**Time Zone**.

2. In the **Value** column, select the time zone from the list.

3. Click out of the field or press **ENTER** to save the change.

QUOTA WARNING

Select the threshold that must be met before the system generates a warning to the subscriber. This value is set as a percentage of the Maximum Mailbox Message Capacity (under Message Limits). For example, if this value is set to 80 and the Maximum Mailbox Message Capacity value is set to 30 minutes, a warning message is issued when the number of voice mail messages totals 24 minutes (80% of 30 minutes). This warning prompt is then played each time the user accesses their mailbox. The warning prompt is no longer played after the mailbox storage total falls below the quota warning threshold. The valid range is 0–100 percent, and the default is 80. If set to 0 or 100, no warning message is issued.

To set the Quota Warning value:

1. Select Voice Processor – Devices – Mailboxes – Local – *<extension>* – **Quota Warning**.

2. In the **Value** column, select the option from the list.

3. Click out of the field or press **ENTER** to save the change.

QUOTA GRACE

Select the amount of additional storage that the system allows once the Maximum Mailbox Message Capacity limit (under Message Limits) is met. This value is also set as a percentage of the Maximum Mailbox Message Capacity. To determine the point at which messages are denied, the system adds the quota grace value to the total capacity limit. For example, if this value is set to 80 and the Maximum Mailbox Message Capacity is set to 30 minutes, the system does not store messages once the number of voice mail messages totals 54 minutes (30 +.8 x 30). When this capacity is reached, a voice mail prompt informs the user that their mailbox is currently full. The valid range is –1 to 100 percent, and the default is 0, which indicates that no grace is allowed. For unlimited messages, set this value to –1.

To set the Quota Grace value:

1. Select Voice Processor – Devices – Mailboxes – Local – *<extension>* –**Quota Grace**.

2. In the **Value** column, select the option from the list.

3. Click out of the field or press **ENTER** to save the change.

LANGUAGE

The Language field determines the language used for mailbox displays, not prompts, such as E-Mail Synchronization.

To set the Language value:

1. Select Voice Processor – Devices – Mailboxes – *<mailbox>* – **Language**.

2. Select the Language drop-down list, and then scroll to the language you want to use. The language choices are:

    - Use Primary Language (default)
    - Use Secondary Language
    - American English
    - British English
    - Canadian French
    - Spanish

3. Press **ENTER** or click outside the field to save the change.

MAILBOX-RELATED INFORMATION

You can view voice processing mailbox statistics. This information is presented for reference only and cannot be programmed. Statistics continue to accumulate until they are cleared using the Voice Processor Report Parameter window. See "Application and Channel Statistics" on **page 1025**) for more information.

To view mailbox statistics:

Select System – **Mailbox-Related Information**. The following statistics are shown in the right pane:

- **Number of Mailboxes**: The number of mailboxes that have been created in voice mail.

- **Number of Messages Sent**: A count of the number of times subscribers have recorded and sent messages to one mailbox or to a group list of mailboxes, replied to a message sent by another subscriber, or forwarded a message with comments.

- **Number of Messages Received**: The number of messages that subscribers have received regardless of the origination of the message—subscriber, non-subscriber, or system.

- **Number of Messages Received From Remote Nodes:** The number of messages that subscribers have received from phones on other nodes.

- **Number of New Messages**: The total of the number of messages in all subscribers' new message queues.

- **Number of Saved Messages**: The total of the number of messages stored in the saved message queues for all mailboxes.

- **Total Length of New and Saved Messages**: A combined total of the amount of time represented by the "Number of New Messages" and "Number of Saved Messages" fields.

- **Average Mailbox Percent Full**: The average percentage of maximum message capacity used by all mailboxes.

- **Number of Times Mailbox Was Full**: The number of times any mailbox reached its maximum message capacity.

- **Number of Times Mailbox Was 80% Full**: The number of times any mailbox reached 80% of its maximum message capacity.

- **Number of Times 3 Bad Passcodes Were Entered**: The number of times that subscribers attempted to use an incorrect password three times in row to connect to a mailbox.

- **Number of Mailboxes Currently Full**: The number of mailboxes that are currently at their maximum capacity.

- **Number of Mailboxes Currently More Than 80% Full**: The number of mailboxes that are currently over 80% of their maximum message capacity.

Chapter 16

VOICE PROCESSING MANAGEMENT

INTRODUCTION

This chapter describes tools that you can use for Unified Voice Messaging (UVM) maintenance and how to save or restore UVM databases. For MiCollab Unified Messaging system maintenance, refer to the *MiCollab Unified Messaging System Administration Help* and the *MiCollab Unified Messaging Technician's Handbook*.

For voice processing-related troubleshooting issues, see .

The type of automatic maintenance performed is up to you. With Windows-based VPU, you can use the Watchdog Configuration Utility (located in the `avdap\utility` directory) to configure automatic backups of the Voice Processor database. You also have the option of creating a batch file to perform this task and/or other custom maintenance tasks. When you create the batch file, name it `system_maintenance.bat` and place it in the `winnt/system32` directory.

An example of a simple batch file that will copy the entire db\ directory to a directory called backup\avdapdb\ is as follows. (Contact Mitel for more complex examples of batch files.)

```
net stop avdap
del /s /f /q c:\backup\avdapdb
rd /s /q c:\backup\avdapdb
md c:\backup\avdapdb
xcopy /s /e /v c:\avdap\db c:\backup\avdapdb
net start avdap
```

With Windows-based version, Voice Processing first checks the Auto Reboot and Auto Maintenance fields.

- If the current day is set for Auto Maintenance, the software watchdog will run the Auto Maintenance batch file. When Auto Maintenance is finished, the software watchdog will check if Auto Reboot is enabled for that day. If so, it will shut down Windows-based VPU and reset the PC. If the current day is not set in the auto reboot field, the software watchdog will just restart the Avdap (Voice Processing) service.

- If the current day is not set in the Auto Maintenance field but it is set in the Auto Reboot field, the software watchdog will shut down both the Avdap service and Windows-based VPU, and reboot the PC.

You can program either Auto Maintenance, Auto Reboot, or both. However, on a day where both fields are programmed, the software watchdog will run Auto Maintenance first and then Auto Reboot.

ENABLING DIAGNOSTICS

When the Enable Diagnostics option is selected, voice processor logs all diagnostics output (including alarms) generated by the voice processor computer to a file. Using that file, you can troubleshoot problems dealing with message lamps, delayed messages, remote messaging, and so forth. By default, this flag is disabled.

There are five diagnostics log files: log, log.1, log.2, log.3 and log.4. Information is initially written to the log file, when this file becomes full its contents are transferred to the log.1 file. The log.1 file contents are transferred down to the log.2 file, whose contents are transferred to the log.3 file, whose contents are transferred to the log.4 file. The information contained in log.4 is discarded permanently each time log.3 is transferred.

Diagnostics may be left on at all times without affecting performance. The number and size of the log files can be changed. For a complete list of diagnostic codes, refer to the *Message Print Diagnostics Manual,* part no. 550.8018, which is supplied on the System software DVD. You can also find all documentation on the Mitel eDocs Web site (http://edocs.mitel.com).

To select the Enable Diagnostics option:

1. Select Voice Processor – Maintenance **– Enable Diagnostics**.

2. In the **Value** column, select the check box. The field changes to **Yes**. To disable the option, clear the check box.

3. Click out of the field or press **ENTER** to save the change.

Voice Processor Database Backup

This section describes the procedure for performing automatic Voice Processor database backups using Windows-based VPU. These backups can be written to a local drive, a network drive, or to an attached mass-media device. For the duration of the backup, the Voice Processor is *not* functional. When using a network drive, you can configure the Voice Processor to backup to a local drive first to minimize downtime.

NOTE	If setting up the system for automatic backups, you must attach a standard USB flash drive to the USB-A port on the front of the chassis. This flash drive must remain attached at all times

To configure the system to perform automatic backups:

1. Start the Watchdog Configuration Utility, and then click the Maintenance tab.

2. Turn on the **Maintenance Enabled** flag (a check mark is displayed).

3. Enter the time of day when the backups will be performed in the Maintenance Time field.

4. Select the days of the week to perform the maintenance in the Maintenance Days field. You can perform Steps **1** through **4** through the DB Programming Maintenance menu (Voice Processor\Maintenance).

5. Set the Temporary Local Save Directory field to the location on your hard drive where the Voice Processor backup will be temporarily stored during the backup operation (for example, `d:\backup`). Although this step is optional, it will help to substantially reduce downtime during backups to mass-media drives. The backup files are copied to the local hard drive,

then copied to the appropriate destination on the mass media device. This allows the Voice Processor to start as soon as the backup to the local drive is complete.

6. Set the Voice Processor Database Save Directory field to the USB flash drive location (where the Voice Processor backup will be stored).

SAVING AND RESTORING VOICE PROCESSING DATABASES

NOTE	Saving or restoring a database causes the system to slow down. If possible, do not perform these operations during normal business hours.

This section contains the following information:

- **Summary of Voice Processor Save and Restore Options** below

- Voice Processor Save and Restore on **page 1002**

- Voice Processor Save Guidelines on **page 1003**

- Save To or Restore To Location on **page 1003**

- Voice Processor Data on **page 1004**

- Completing the Save/Restore on **page 1005**

- Enabling/Disabling a Voice Processor on **page 1006**

- Saving a Voice Processor Database in Online Mode on **page 1007**

- Selecting a Voice Processor Type in Offline Mode on **page 1007**

- Saving/Restoring Data on **page 1008**

- Saving/Restoring Data in a System Equipped with a PS-1 on a Remote PC on **page 1012**

- Saving/Restoring UVM Data in a System Equipped with a PS-1 on **page 1016**

- Saving/Restoring UVM Data to a NFS-Supported Computer on **page 1016**

SUMMARY OF VOICE PROCESSOR SAVE AND RESTORE OPTIONS

With systems not quipped with a PS-1, you can save or restore Unified Voice Messaging (UVM) databases using the USB-A drive on the front of the Base Server chassis only. For systems equipped with a PS-1, UVM databases are stored on the hard drive.

You can save and restore UVM data to a USB flash drive. For systems equipped with a PS-1 and licensed to run 16-port UVM, you can save/restore UVM data to a remote computer after setting up a shared folder on the computer.

You can save/restore UVM to a remote computer running Windows 2000 or later or to a Linux/Unix Network File System (NFS) shared folder.

NOTE	Before you can save/restore UVM data to a remote Windows or Linux/Unix computer, you must set up a shared folder on the remote computer.
	For detailed instructions as applicable, see "Saving/Restoring Data in a System Equipped with a PS-1 on a Remote PC" on **page 1012** or "Saving/Restoring UVM Data to a NFS-Supported Computer" on **page 1016**.

Table 165 shows the options available for saving and restoring voice processor databases.

Table 165:   Voice Processor Database Save/Restore Options

SYSTEM	WITH UVM
Without PS-1	To USB
With PS-1	To USB or Remote Windows or Remote NFS

NOTE	You can use save or restore all fields except the Call Processing Server password. Also, an error message appears if you attempt to restore an incompatible system type onto a system. For example, you cannot restore a Base Server database onto a PS-1 system and vice versa.

VOICE PROCESSOR SAVE AND RESTORE

The selective Voice Processor Save and Restore options are available under the Operations menu.
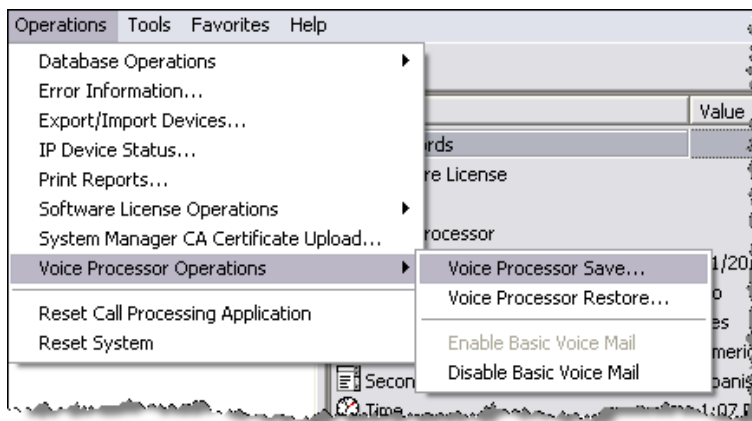
When you select either Voice Processor Save or Voice Processor Restore from the Operations – Voice Processor Operations submenu, one of the following window appears. When using Unified Voice Messaging, the SMDR Buffer and Fax Document options are not available.
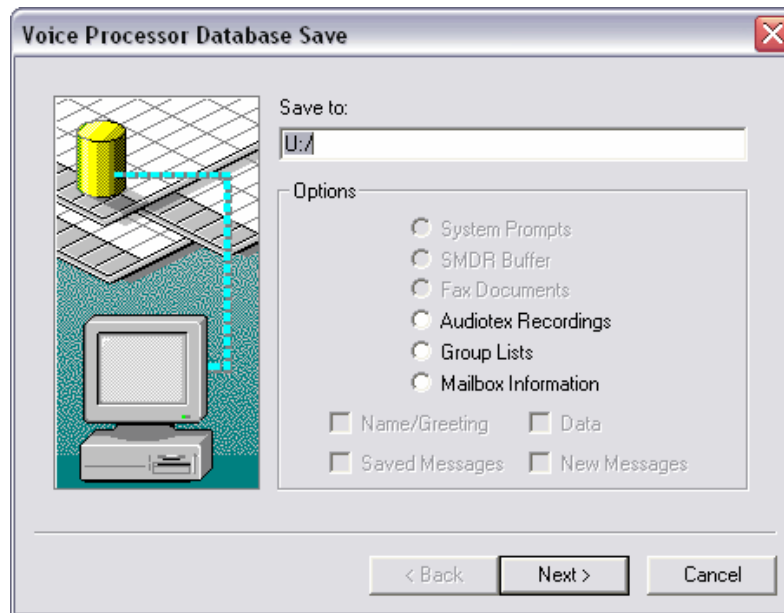
VOICE PROCESSOR SAVE GUIDELINES

Keep the following guidelines in mind when saving voice processor information:

- If you are using an external voice processing system, make sure the system is attached during a restore. If the external voice processing system is not attached, the system will not restore the non-associated mailbox information.

- If saving to a USB flash drive, make sure the flash drive is inserted on the MiVoice Office 250 Base Server *before* continuing with any voice processing database or restore operations. Create any specific directories *only while the USB flash drive is inserted into the MiVoice Office 250 Base Server chassis*. After performing the save operation, remove the USB flash drive, then reinsert it to verify the information in the directories you created. You will need to repeat the save operation for each subdirectory created.

- If saving external voice processing information to floppy diskettes, ensure you have enough diskettes for saving the data. Check the Disk Usage Statistics report (as shown on **page 863**) to determine the number of minutes used for the prompts and messages. Each diskette will typically hold around 5 or 6 minutes of prompts and/or messages. To calculate the number of diskettes needed, divide the number of minutes used (shown in the statistics) by 5. If applicable, use a mapped drive or DVD-ROM drive instead of floppy disks.

SAVE TO OR RESTORE TO LOCATION

If saving or restoring to a USB flash drive (Unified Voice Messaging only), `U:\` appears in the Save To/Restore From field. If using an external voice processing PC, set the Save to and Restore from fields accordingly. For example, `A:\<path>` if using floppies or `D:\` if using DVD-ROM or other drive where the voice data was saved to or will be restored from.

NOTES	Always use the option to save voice data if you are defaulting the database or changing the flash card, as this does a complete save of the system and voice information. When you then do a restore, all messages are restored as saved and new respectively. If you simply save the voice data then default the database or change the flash card, or remove the messages through the database or the phone, the restored voice data will be restored as new.

Selecting the desired drive places a dot in the option button. Place a diskette in the correct drive on the voice processor computer before selecting **Next**. If you are using the USB flash drive for UVM, make sure the drive is inserted into the MiVoice Office 250 before specifying the save to or restore from options.

> ## NOTICE
>
> **Possible Database Loss and VM System Inoperability**. When an external voice processing system performs a save operation, it erases files and/or directories from the UNC_path. Therefore, be careful when specifying the save/restore path. For example, do not specify C:\ as the save/restore path because the external voice processing system would delete the entire C drive.

VOICE PROCESSOR DATA

Select information you want to save or restore from the following list. If you select Mailbox Information, you will also need to select the types of information by clicking the appropriate check boxes. Restoring Voice Processor data will not allow you to create new mailboxes or group lists. You can only restore data to mailboxes and groups lists that already exist in the System database.

NOTE	When using a USB flash drive to back up Unified Voice Messaging, and you want to back up specific voice data, create subdirectories on the flash drive while it is in the MiVoice Office 250 Base Server. For example, to back up prompts, create a subdirectory at the root level of the flash drive such as U:\prompts. Specify this as the Save to location in the dialog box shown in the figure on the previous page. Repeat the save to process for each subdirectory.

- **System Prompts:** The system prompts (default and customized) will be saved or restored. Restored information will overwrite the current prompts, if any.

- **SMDR Buffer:** If you select Save, the current call record information in the SMDR buffer will be saved to the selected drive. The buffer will not be cleared. If you select Restore, the SMDR call records contained on the disk in the selected drive will overwrite the existing SMDR buffer contents.

- **Fax Documents:** All fax documents in the database will be saved or restored. Restored documents will overwrite existing documents of the same number, if any.

- **Audiotex Recordings:** All audiotex recordings will be saved or restored. Restored recordings will overwrite the current recordings of the same number, if any.

- **Group Lists:** All group lists and group list directory information in the database will be saved or restored. Restored information will overwrite existing information for group lists of the same number, if any.

- **Mailbox Information:** You can select the mailboxes information to be included. Depending on the items checked, the following will be saved or restored:

- **Name/Greeting:** Name and greetings recorded by the Subscribers, including primary and alternate greetings and directory names for extension IDs.

- **Saved Messages:** Messages that have been saved by the Subscribers.

- **Data:** User statistics for each mailbox and extension ID.

- **New Messages:** New and undelivered messages that have not been heard by the Subscribers.

NOTES	Restored mailbox information overwrites existing information in the database. If any certified messages are restored, the "certified" flag will be cleared so that the sender does not receive duplicate receipt messages. If a message being restored is already in the mailbox, the message will not be copied from the disk. It will not create a duplicate copy of the message. Restored messages appear labeled as "New" after the restore has completed.

COMPLETING THE SAVE/RESTORE

Selecting Recordings, Group Lists, or Mailboxes: If you are saving recordings, group lists, or mailbox information, you must select the items you want to save. When you click **Next**, a window appears that allows you to select the items that you want to save. You can select several consecutive items by selecting one, holding down **SHIFT** and then selecting the last. Or you can select individual items by holding down **CTRL** while clicking on the desired items. When you have finished, click **Next** again.

Next/Back: When you click **Next**, the selected information is displayed for reference. If it is not correct, select **Back** and correct the information. If it is correct, click **Finish** to continue.

Finish: When you click **Finish**, the information will be saved or restored to the selected drive. The following windows may appear:

- A restore operation begins with a warning that the database is about to be overwritten with the new data. You are given the choice of allowing the overwrite (select **Yes**) or canceling the restore process (select **No**).

  If using a USB flash drive to back up Unified Voice Messaging, and you want to back up specific voice data, create subdirectories on the flash drive while it is in the MiVoice Office 250 Base Server. For example, to back up prompts, create a subdirectory at the root level of the flash drive such as U:\prompts. Specify this as the Save to location in the dialog box in the figure on page 1003. Perform the save operation for each subdirectory.

  If using diskettes with an external voice processing system, use a separate disk for each type of save (System Prompts, SMDR Buffer, and Fax Documents). Each time you perform a save operation, *all files* on the disk are erased and replaced with the new saved information.

- During a save or restore process, a window appears that shows the percentage of the database that has been copied. To cancel the save or restore operation, select **Cancel**.

NOTE	Canceling a restore operation will cause the system to default the Voice Processor database.

- If restoring mailbox information, group lists, or audiotex messages, place the *last* disk of the saved information in the drive. This disk has the file that tells the system which information was saved and has the requested information included in the saved data. The system checks to see if the requested information is contained on the disks. If not, you will see an

error message. If the information is available, you will be instructed to insert the appropriate disk(s) to complete the restore operation. Only the information you checked in the Save/ Restore windows will be restored to the system, even if other information is present on the disks.

The Voice Processor applications and/or mailboxes will not be usable while they are being saved or restored and callers will receive reorder tones. This is necessary to prevent users from making new recordings and causing database errors. *If a mailbox is busy when you attempt to restore it,* you will receive a warning message with the following three options:

- *Abort the Save/Restore operation:* This terminates the entire save or restore operation, even if you were performing the operation for more than one mailbox.

- *Cancel the operation for that mailbox:* This terminates the save or restore operation for the busy mailbox.

- *Try again:* If you are saving or restoring multiple mailboxes, it places the busy mailbox at the end of the list and retries it later. If only one mailbox is being restored, it immediately tries again. If the mailbox is still busy when retried, you receive the three options again.

ENABLING/DISABLING A VOICE PROCESSOR

You can enable or disable the voice processing application.

To disable voice processing:

1. Back up the Voice Mail database. You need a USB flash drive for this operation if backing up Unified Voice Messaging. See "Voice Processor Save and Restore" on **page 1002**.

2. Launch System Administration & Diagnostics, and then start a DB Programming session.

3. Select the **System** folder, and then click **Enable Voice Processor**.

4. Click the **Yes** value, and then click in the check box to remove the check mark. Click outside the area to implement the change. Click **OK** to the prompt that appears. DB Programming restarts.

The Enable Voice Processor database field is autoenabled by Call Processing. For example, if the Enable Voice Processor flag is set to No, and the user successfully connects an external voice processing system or enables Unified Voice Messaging, Call Processing auto-enables the flag. However, if DB Programming is currently programming the switch, the auto-enable flag does *not* get automatically enabled.

NOTE	In its default state, this flag is turned ON. If you do not have an external voice processing system connected to the MiVoice Office 250 Base Server, make sure to turn this flag OFF before attempting to import or export information over the network.

SAVING A VOICE PROCESSOR DATABASE IN ONLINE MODE

You can save UVM data in online mode. For proper configuration, DB Programming requires that the voice processor type is identified. After the voice processor is physically connected, the system senses which type it is.

A voice processor connected to a MiVoice Office 250 auto-enables when the system boots up. To avoid the risk of overwriting an established database with default values, Mitel recommends connecting the voice processor first, then connecting with an online mode session of DB Programming, and then saving.

To save a voice processor database in online mode:

1. Connect the PS-1 equipment to the system in accordance with installation instructions. Refer to the applicable installation manual.

2. In online mode, launch System Administration & Diagnostics, and then start a DB Programming session.

3. From the Mitel DB Programming menu bar, select Operations – Save/Restore Node Data – **Database Save**. Identify the location where you want to store the system database, then select the Save Voice Data to: check box and identify the location where you want to save the voice processor database.

4. Click **Start**. System and voice processor databases are saved to the locations indicated.

SELECTING A VOICE PROCESSOR TYPE IN OFFLINE MODE

When you enable a voice processor in offline mode, you may be prompted to choose a voice processor type. The Select a Voice Processor Type dialog appears *only* if you have saved the database from a remote session that did *not* have the voice processor physically connected. For the recommended process, see "Saving a Voice Processor Database in Online Mode" on **page 1007**.

The Select a Voice Processor Type dialog allows you to identify a known type of voice processor that has not been enabled in the system configuration. It does *not*, however, allow you to change a voice processor type after the database has been created.

To enable a voice processor in offline mode:

1. Launch System Administration & Diagnostics, and then start a DB Programming session.

2. From the left pane of the Mitel DB Programming main screen, select the **System** folder. Contents of the System folder appear in the right pane.

3. Click the value of the Enable Voice Processor feature to **Yes**.

NOTE	The "Select a Voice Processor Type" dialog appears only if you have, from a remote session, saved the database of a system that does *not* have a voice processor unit physically connected. Once the voice processing unit is connected, the applicable function enabled, and a database saved in a remote session, the Select a Voice Processor Type dialog does *not* display again.

4. Select the option button for the voice processor type you know is connected to the system.

5. Click **OK** to continue programming the selected voice processor. If you select **Cancel**, the Enable Voice Processor field in the System folder toggles to **No.** If you select a voice processor type, or if one already exists in the database, you must restart the system in order for the changes to take effect. The following message window appears.
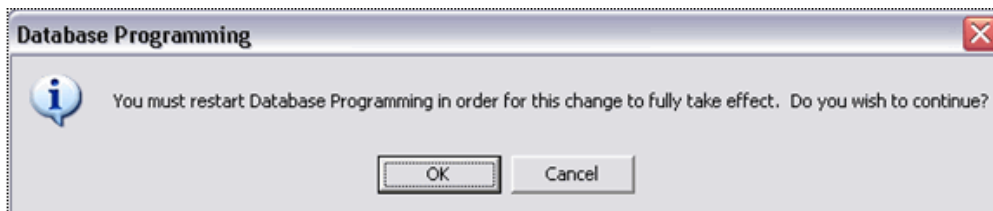
6. Click **OK.** The voice processor type is updated in DB Programming and the session closes. Clicking **Cancel** toggles the Enable Voice Processor value in the System folder to **No,** cancels any Select a Voice Processor Type selection made, and displays the Mitel DB Programming window.

7. Restart a local DB Programming session.

8. After enabled, the voice processor can be configured and programmed in accordance with customer requirements.

SAVING/RESTORING DATA

The following instructions explain how to save or restore voice processor data on a system equipped with a PS-1. For detailed instructions about saving and restoring voice data on a system that does not have a PS-1 and is running UVM, refer to the *MiVoice Office 250 DB Programming Help.*

Save/restore options appear in the dialog box on the first page of the Voice Processor Save/Restore wizard.

From the dialog box similar to the one in the following example, you may select only Voice Mail Drive. Because you are not connected to a system equipped with a PS-1 and supporting UVM, the Windows and NFS options are dimmed and *not* selectable.
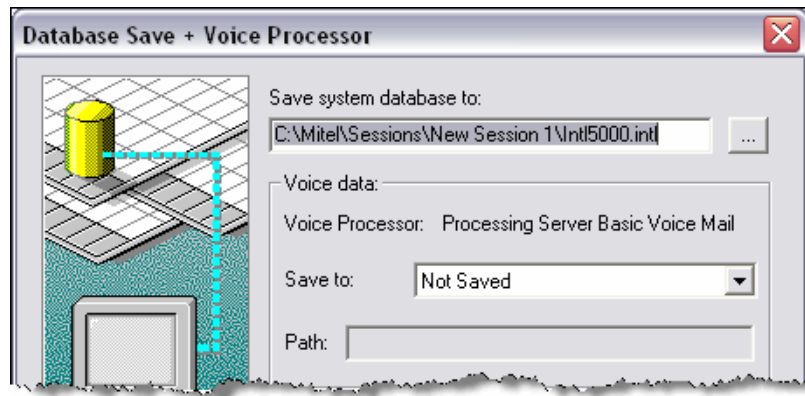
In the Path box, enter the drive letter and path to the location of the local unit.

If a system that is equipped with a PS-1 and running UVM is connected and you are saving to or restoring from a Windows or NFS computer, the Hostname field is enabled. The Hostname field is dimmed and unselectable when you select a USB location.

The Hostname field and the path to it are filtered by DB Programming to ensure that only valid characters are entered. For the Hostname, a valid number or letter must be entered as the starting character. The remaining characters can be any valid character for a hostname.

Table 166 shows valid characters for both path and Hostname entries.

Table 166:   Valid Characters for Path and Hostname Entries

FIELD	DEFAULT	RANGE
Hostname	Blank	0–63 characters
		Valid characters for first character: 0–9, a–z, A–Z
		Valid characters for remaining characters: 0–9, a-z, A–Z, '-', and '.'
Path (Character validation applies to all system types.)	For external Voice Mail drive selection, E:\ For Unified Voice Messaging drive selection, U:\	1–65 characters All characters are considered valid except the following: * . ( ) , | ` ; ? < > \ "& $ ^ % # In Windows and NFS, the path should start with a single backslash (\). Two backslash characters to start the path should work as well, but three invalidate the path.

For a Windows or NFS location, the Hostname field is enabled. Enter a hostname of a PC on the network for the path in the **Hostname** field. The Hostname field is disabled for a USB location selection. The Hostname field is filtered by DB Programming to ensure that only valid characters are entered. A number or letter must be entered as the starting character. The remaining characters can be any valid character for a hostname. Table 10 summarizes the valid characters. If you do not specify a hostname, the following warning appears

Remember the following when restoring a database:

- If you attempt to restore an incompatible system type onto a system, an error message appears. For instance, you cannot restore a Base Server database onto a system equipped with a PS-1 and vice versa.

- Restoring a database with more IP Devices than is supported in the system is *not* allowed. If this illegal operation is attempted during a Database Restore procedure, a warning message appears, and the restore is canceled.

- When restoring a database, if the IP settings on the Base Server differ from the settings in the database to be restored, a dialog similar to the following one below appears. Previously, the SSH Server Enabled and Web Server Enabled fields were available on this dialog so that the current values for these fields could be preserved. The option to preserve the current value for these fields is no longer included in the dialog. Only fields pertaining to the system identity on the network are provided.

Database Restore

The current system Base Server IP Settings differ from the settings in the database being restored. Please click a button to select the Base Server IP Settings the system will retain.

[Current Settings] [Restore Settings]

Field	Current Value	Restore Value
Static IP Address	192.168.200.201	172.15.14.232
Static Subnet Mask	255.255.255.0	255.255.255.0
Static Gateway	192.168.200.1	172.15.14.1
Hostname	JBS-IT5K	JBS-IT5K
DHCP Enabled	Yes	No
Base Server/Processing Server Connection Address	172.17.158.89	172.15.14.232
Base Server/Processing Server Connection Port	3400	3200

To keep the current IP settings, click **Current Settings**. The settings in the Current Value column are copied over to the database.

To restore the IP settings, click **Restore Settings**. The settings in the Restore Value column are copied over to the database.

- For systems equipped with a PEC-1, if the current settings for the PEC-1 differ than the settings in the database to restore, the following dialog box appears, allowing you to specify which settings you want for the Expansion card of your system.
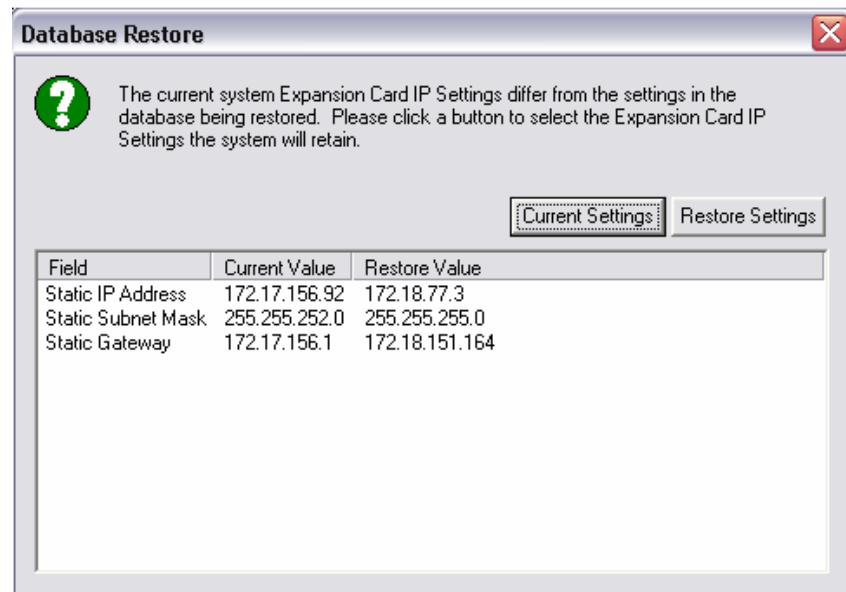
- For systems equipped with a PS-1, the following dialog box displays if the system PS-1 settings differ from the database to be restored.
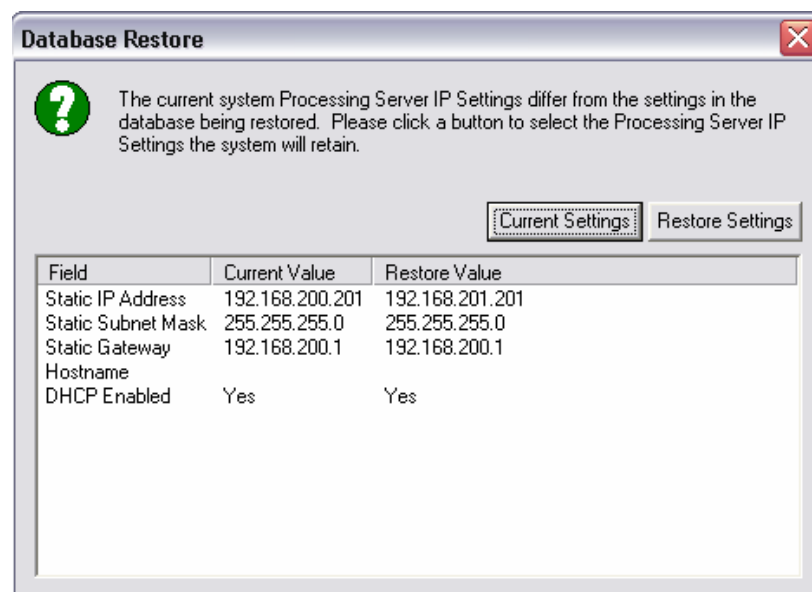
To save or restore voice data on a system equipped with a PS-1 to a local drive:

1. *In online mode and connected to a system equipped with a PS-1.* From the DB Programming menu bar, select Operations – Save/Restore Node Data – **Database Save…** (or **Database Restore**). A dialog box similar to the previous one appears.

2. Select the **Save voice data to:** or **Restore Voice Processor voice data:** option. The **Voice Mail Drive** option is automatically selected.

3. In the Path box, type the path (up to 256 characters). If you are saving to or restoring from a voice mail drive, **E:** appears in the Path: box. If using an external voice processor computer, browse to the applicable destination drive and folder sequence where the voice data

was saved to or is restored from. When a save or restore path is browsed or used, DB Programming remembers the path last used, per PC user. Valid characters appear in **Table 166**. If you do not specify a path, the following message window appears.

4. Click **Start**.

SAVING/RESTORING DATA IN A SYSTEM EQUIPPED WITH A PS-1 ON A REMOTE PC

To save or restore UVM data from the system equipped with a PS-1 to a computer running Windows 2000 or later, the destination computer must have a "Guest" account enabled and must be allowed to share folders.

Depending on the Windows operating system and how the computer is configured, the procedure may differ from the one shown in the following example. However, the setup procedure needs to be done only once for the destination computer. The following procedure uses examples from Windows 2000 user interfaces and can be performed only by a user with administrative privileges.

NOTICE

Security Concern. Using a shared folder on a remote computer for saving and restoring UVM data may introduce an undesirable network security risk. Before setting up a shared folder on a remote computer for this purpose, Mitel recommends that you obtain permission from the network administrator.

The shared folder should be considered a temporary location for the purposes of the backup only. Once the backup has completed, you should move the data to a secured folder.

To identify active policy settings on Windows 2000 computer:

1. From the Windows task bar, select Start – Control Panel – **Users and Passwords**.

2. In the Tree tab in the left pane of the Local Security Policy Settings window, select Security Settings – Local Policies – **User Rights Assignment**. The Local Security Settings window displays.

3. In the Policy column, double-click the policy titled, "Access this computer from the network."
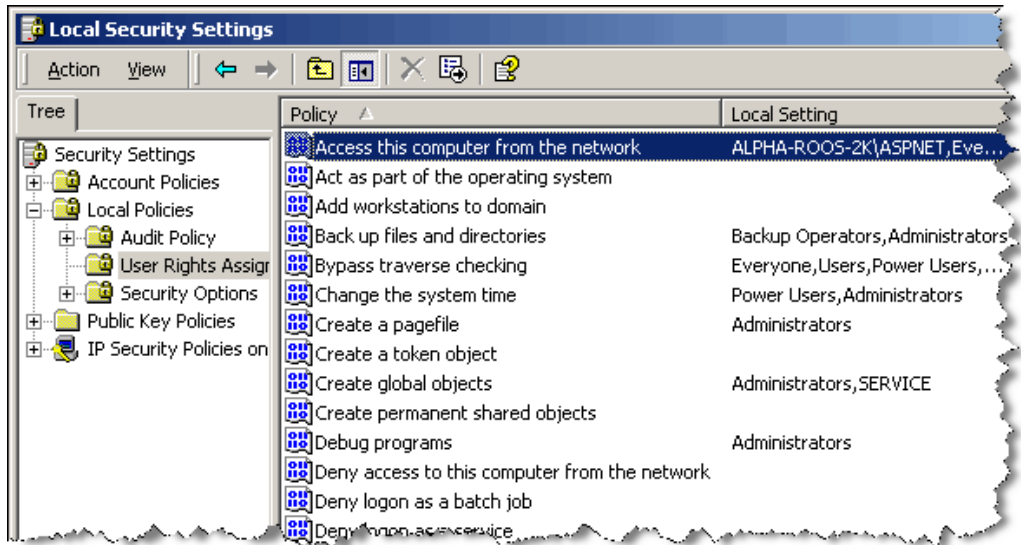
The Local Security Policy Setting window appears, showing a summary of active local policy settings.
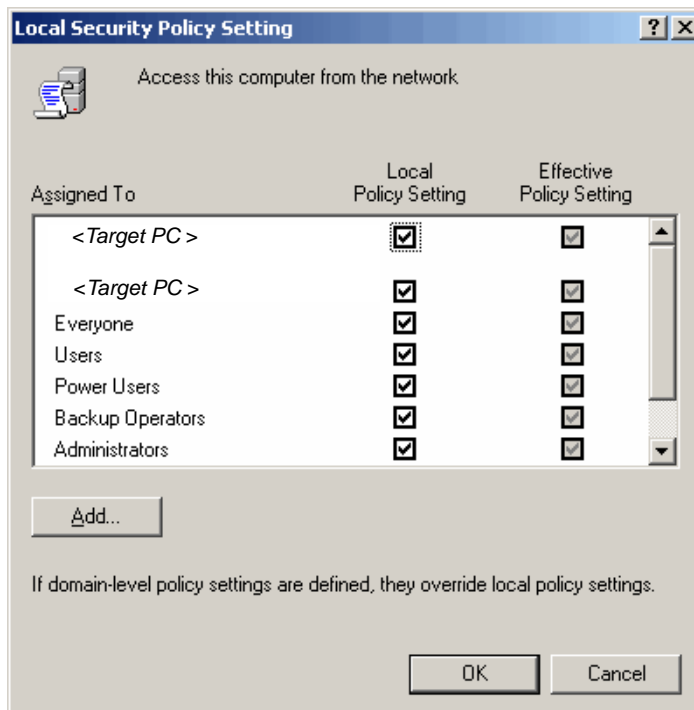
4. Verify that Guest appears on the list and is checked. If missing, add Guest to the Local Security Policy Setting list.

To add Guest to the Local Security Policy list:

1. In the Local Security Policy Setting window (see the preceding procedure), click **Add**. The Select Users or Groups window displays.

NOTE	Be aware that a user group exists in Windows that is named Guests, with an "s" at the end. Be careful not to confuse this similar name with the singular user name, Guest.

2. In the list box find the account for Guest in the Name column. The value in the In Folder column must be the name of the destination computer. Make sure the name of the destination computer appears in the Look in list.
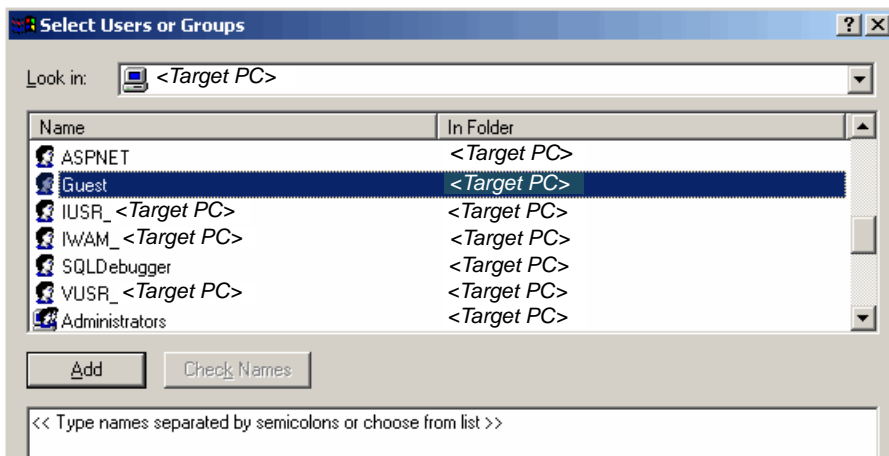
3. Select **Guest**.

4. Click **Add**, and then click **OK**.

5. Make sure that Guest does not appear on the Local Security Settings policy titled, "Deny access to this computer from the network."
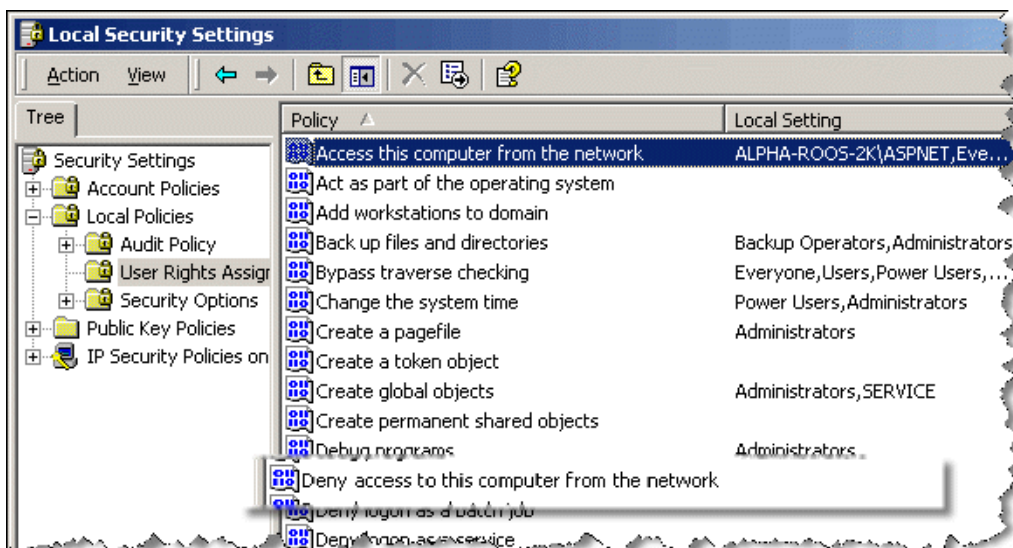
From Windows Explorer, create and name an empty folder into which you will save the system UVM database and from which you will restore the database to the PS-1. In the following example, the name of the folder is "share."

To add a shared folder on a Windows 2000 computer:

1. From Windows Explorer, right-click the empty folder you created in the preceding step and click **Properties**. The "share Properties" window displays.

2. If Guest does not appear on the user list, under the Sharing tab, click **Permissions**, and then click **Add**. To enable the Guest account to access the shared folder, make sure the Location is set to the name of the destination, or target, computer and type "Guest" in the box.

3. Click **OK**. This action returns you to the "Permissions for share" window.

4. On the Share Permissions tab under Name, select the Guest user with the destination, or target remote computer identified in parentheses. In the Permissions: list box, check Full Control in the Allow column, and then click **OK**.
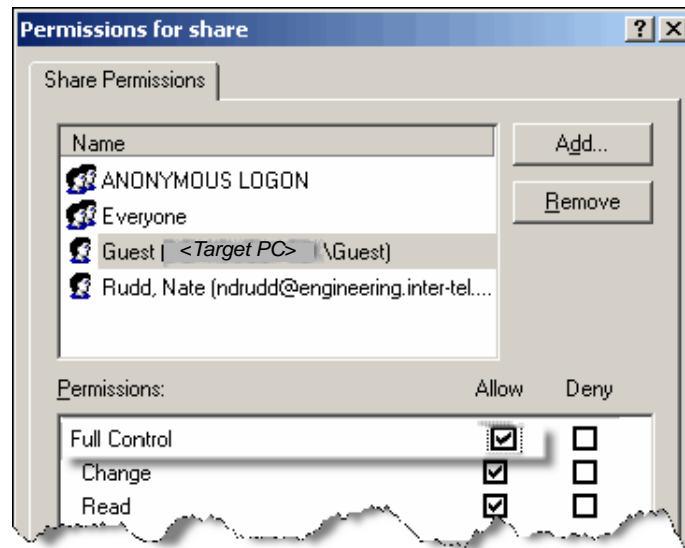
Repeat the preceding procedure on the Security tab in folder Properties—right-click the folder and then click **Properties**. When the Security tab settings have been made, the destination

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folder is ready to be saved to or restored from. In the preceding example, the folder would be accessed by Voice Processor save/restore by providing the hostname <Target computer Name> and the path /share.

SAVING/RESTORING UVM DATA IN A SYSTEM EQUIPPED WITH A PS-1

If the system is equipped with a PS-1 and appropriately licensed for UVM, the Windows and NFS options appear in the Database Save – Voice Processor window.

To save/restore a UVM database to a remote computer Running Windows:

1. Set up an empty shared folder on the remote/destination computer. See the preceding procedure, "Saving/Restoring Data in a System Equipped with a PS-1 on a Remote PC" on **page 1012**.

2. From the remote/destination computer with the shared folder you created in step **1**, launch System Administration & Diagnostics, and then start a DB Programming session.

3. From the DB Programming menu bar, select Operations – Voice Processor Operations – **Voice Processor Save** or **Voice Processor Restore**, as appropriate.

4. Select the Save to check box.

5. Select the applicable destination, and then click **Next**.

6. In the Path box, type the path on the remote/destination computer to the shared folder created in step **1**.

7. In the Hostname box, either enter the path and name of the remote/destination computer on the network or enter the IP address of the destination computer.

8. Click **Start**.

SAVING/RESTORING UVM DATA TO A NFS-SUPPORTED COMPUTER
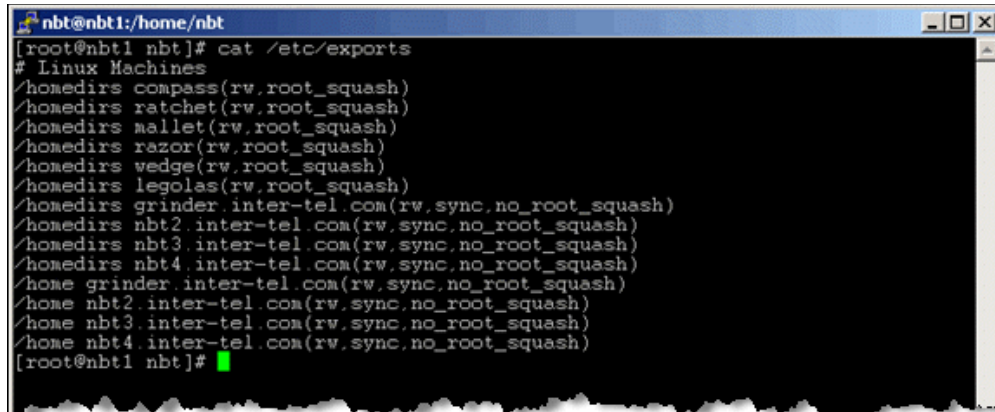
This procedure typically applies when using a computer running a Linux operating system, although it would also apply to a Unix-driven computer. Settings may vary because of the many distributions of Linux. You must have root privileges to set up the system for saving or restoring UVM data.

For online assistance to configure Network File Server (NFS), go to http://nfs.sourceforge.net/nfs-howto/.

The nfsd process must be running and the folder you want to save must be exported to the IP address of the PS-1. In the Red Hat/Fedora flavors of Linux, you can find the export table listed in the file /etc/exports.

<div style="background-color:green; color:white">

NOTICE

</div>

Security Concern. Using a shared folder on a remote computer for saving and restoring UVM data may introduce an undesirable network security risk. Before setting up a shared folder on a remote computer for this purpose, Mitel recommends that you obtain concurrence from the customer's LAN administrator.

```
nbt@nbt1:/home/nbt                                              _ □ ×
[root@nbt1 nbt]# cat /etc/exports
# Linux Machines
/homedirs compass(rw,root_squash)
/homedirs ratchet(rw,root_squash)
/homedirs mallet(rw,root_squash)
/homedirs razor(rw,root_squash)
/homedirs wedge(rw,root_squash)
/homedirs legolas(rw,root_squash)
/homedirs grinder.inter-tel.com(rw,sync,no_root_squash)
/homedirs nbt2.inter-tel.com(rw,sync,no_root_squash)
/homedirs nbt3.inter-tel.com(rw,sync,no_root_squash)
/homedirs nbt4.inter-tel.com(rw,sync,no_root_squash)
/home grinder.inter-tel.com(rw,sync,no_root_squash)
/home nbt2.inter-tel.com(rw,sync,no_root_squash)
/home nbt3.inter-tel.com(rw,sync,no_root_squash)
/home nbt4.inter-tel.com(rw,sync,no_root_squash)
[root@nbt1 nbt]# █
```

To set up an NFS server to Save/Restore voice processing data:

1. Add the absolute path (starting from /) to the folder you want to export and then the IP address or name of the PS-1 server you are exporting to. Example: If the IP address of the PS-1 is 155.166.1.1 and you want to share a folder named `/vp_data`, you would add the following line to the exports file:

   `/vp_data 155.166.1.1(rw,sync,no_root_squash)`

2. After editing this file, execute the command `exportfs` and restart the `nfsd` process.

To save/restore a UVM database to a remote computer Running Linux/Unix NFS:

1. Set up an empty shared folder on the remote/destination computer. See the preceding procedure.

2. From the remote/destination computer with the shared folder you created in **step 1**, start an nfsd process.

3. From the Mitel DB Programming menu bar, select Operations – Voice Processor Operations – **Voice Processor Save** or **Voice Processor Restore**, as appropriate.

4. Select the Save voice data to check box.

5. Select the NFS option, and then click **Start**.

TROUBLESHOOTING A VOICE PROCESSING

Table 167 summarizes the troubleshooting strategies recommended for resolving discrepancies with voice processing systems.

Table 167:   Voice Processing Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Phone system unable to communicate over voice channels with voice processing computer	Programming error	Make sure the appropriate time slot group (number of voice processing voice channels) is assigned to each voice processing application. Voice channels are used for processing calls between the phone system and the voice processing computer. See **"Time Slot Groups"** on **page 868**.
	Loose or defective cabling between the phone system and the voice processing computer.	Check the cabling connecting the system to the computer.
System programming computer unable to communicate with voice processing computer (the monitor may display a message about being unable to communicate with or losing its connection with the voice processing computer)	Programming error	External voice processing programming cannot be accessed until a voice processing communications port has been established. Make sure Unified Voice Messaging is disabled and enable the Voice Processor Connection flag.
	Defective processor module	Replace the defective module.
	The ASAI connection is bad.	The ASAI connection comes from the IP network. Verify the network configuration settings in DB programming and the Avdap Configuration Utility.
Unified Voice Messaging files are corrupted.	Not running the most current software release.	Upgrade the system files to the latest release. If UVM files are still corrupted, refer to the Knowledge Base for additional information. Contact Technical Support for assistance, if needed.
Monitor and keyboard unable to communicate with voice processing computer	Loose or defective cabling between the voice processing computer and the monitor or keyboard	Check the cabling connecting the computer to the monitor or keyboard.
	Monitor is wrong type for the Monitor Card installed in the computer	Make sure the monitor is the correct type (generally VGA) for the Monitor Card installed in the computer.
	Defective monitor or keyboard	Replace the defective monitor or keyboard.
	Defective Monitor Card or computer Motherboard	Replace the defective card.

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Table 167:   Voice Processing Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Automated attendant not responding properly	User error	The caller must use a DTMF phone to enter digits.
	DTMF tones not being interpreted correctly (see NOTE below)	If numbers appear to be misdialed frequently, due to trunk noise or other problems, use the Digit Translation feature. See "Digit Translation" on **page 884**.
	Programming error	Automated attendant phone(s) must be designated in the database. See "Voice Processing Application Options" on **page 872**.
	Defective VPC or hard disk	Replace the defective computer component.

Note: Due to the natural characteristics of the CO trunk, the volume level of DTMF tones transmitted over the trunk may be substantially reduced before reaching the system. This natural degradation in tone volume may adversely affect the reliability of the automated attendant feature. Other factors which can affect automated attendant performance are CO trunk noise, the quality of the recording device, and the quality and strength of the DTMF tones generated by the calling phone itself.

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Automated Attendant is not receiving incoming calls. Callers are routed to primary attendant instead	Programming error	Ensure that the desired trunks (in day and/or night mode) are programmed to ring in directly to the correct automated attendant extension number. See "Trunk Programming Options" on **page 236**.
	Loose or defective cabling between the phone system and voice processing computer	Check the cabling connecting the system to the computer.
Automated Attendant says you have pressed an invalid button when no button was pressed	Talk off is occurring from the recording	Remove that recording and play the application to see if it transfers improperly. Re-record the message that is causing talk-off.
Automated Attendant indicates that an extension number is invalid even though it exists on the system	The extension does not have an assigned mailbox or an extension ID number	Extension ID numbers provide the Automated Attendant application a means for transferring calls to extensions which do not have mailboxes. See "Extension IDs" on **page 920**.
Automated Attendant transfers calls directly to voice mail without trying the extension number first	Programming error	Make sure the mailbox is associated with a corresponding extension number. See "Programming Mailbox Options" on **page 968**.
Unable to access desired voice processing application (intercom callers hear reorder tones and see a CALL CANNOT BE COMPLETED display; outside callers are rerouted to the primary attendant)	Voice processing has slowed down and call processing has timed out	If the problem continues to occur, even during periods of low activity, reboot the voice processing computer.

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Table 167: Voice Processing Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Unable to access desired voice processing application (intercom callers see a XXXXX IS BUSY display and camp on until a voice channel is available; outside callers hear ringing until a voice channel is available)	All programmed voice channels are currently busy	Make sure the appropriate time slot group (number of voice processing voice channels) is assigned to each voice processing application. Voice channels are used for processing calls between the phone system and the voice processing computer. See **"Time Slot Groups"** on **page 868**.
		Disable Unified Voice Messaging and add an external voice processing application for additional ports.
Unable to access desired voice processing application (intercom callers hear reorder tones and see a XXXXX IS UNAVAILABLE display; outside callers are rerouted to the primary attendant)	A database save or restore is in progress, the computer is being reset, or the computer is starting up for the first time	Try again within five or 10 minutes.
Unable to access desired voice processing application (intercom callers hear reorder tones and see a XXXXX IS UNPLUGGED display; outside callers are rerouted to the primary attendant)	The voice processing computer power switch is off, the cabling between the system and the computer is disconnected or defective, or the voice processing computer is inoperative	Check to make sure the voice processing computer power switch is on, and check the cabling connecting the phone system to the computer. Also, verify IP addressing and network connectivity for ASAI connection.
	The Unified Voice Messaging Application got hung up	Reset call processing.
Voice processing applications are slow (for example, voice prompts are slow and display messages are delayed)	The voice processing computer high-speed (turbo) processor mode is turned off	For optimal performance, the voice processing computer is configured with the high-speed (turbo) mode *always* enabled. Do *not* disable the turbo mode setting.
Voice processing computer is programmed with 12–40 voice channels, but no more than eight calls at a time are accepted	The number of programmed voice channels is less than the available hardware	Make sure the appropriate time slot group (number of voice processing voice channels) is assigned to each voice processing application. See **"Time Slot Groups"** on **page 868**.
	Loose/defective cabling between the voice processing device AIC and the chassis	Check the AIC-to-MiVoice Office 250 Base Server cabling.

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Table 167:   Voice Processing Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
When power is turned on, the voice processing computer does not boot up properly	Diskette in floppy drive	Ensure that the computer is not trying to boot off a diskette in the floppy drive.
	The two-pin AIC-to-reset switch cable is not connected properly	Be sure that pin 1 marked on each end of the cable matches pin 1 on the AIC and computer Mother board. (If connected incorrectly, the computer will be reset continuously or will not boot up at all.)
	The voice processing computer's high-speed (turbo) processor mode is turned off	The voice processing computer boots up much faster in turbo mode. Before shipping, the computer is configured with the high-speed (turbo) mode *always* enabled. Do *not* disable the turbo mode setting.
When power is turned on, the voice processing computer beeps once (for two seconds) and does not boot up properly.	Incorrect DIP switch or jumper strap settings on the Audio Interface Card (AIC)	Ensure that the AIC DIP switches and jumper straps are set in the correct positions.
	Software problem	The computer should be used for voice processing applications only. If necessary, delete any unneeded peripheral software and re-install the voice processing applications software as outlined in the Installation chapter in the *MiVoice Office 250 Installation Manual* .
	Incorrect CMOS settings	Refer to the diagnostic materials generally included in a plastic sleeve attached to the inside of the chassis cover.
Both internal and remote users have no message notification after receiving voice messages in their mailboxes	The Message Notification/ Retrieval application has not been set up	The system must be programmed with the Message Notification/Retrieval application in order to allow voice mail message notification and quick mailbox access. See "Message Notification/Retrieval" on **page 890**.
Audio volume levels from voice processing applications (such as voice mail and Automated Attendant) are too low or too high	Programming error	Adjust the volume level for all of the voice channels used by the voice processing computer. See "Voice Processor System Settings" on **page 862**.
No audio from voice processing applications, such as voice mail or Automated Attendant	Loose or defective cabling between the AIC and the chassis	Check the cabling.

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Table 167:   Voice Processing Troubleshooting Tips (continued)

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
	Invalid or missing prompt files One or more ports are blocked	When Unified Voice Messaging starts up, if the prompt or menu files (english.pmt, english.ini, & menuinit.dat) are not valid (missing, defective, or the wrong version), the message "Faking an active language" will appear in the log file. You must reinstall the prompt and menu files as described in the Installation chapter in the *MiVoice Office 250 Installation Manual* .
	Improper drivers are loaded	Load the correct drivers. See if the fax ports are listed in the summary.txt (in the Avdap folder).
Fax-On-Demand feature not working properly	Programming error.	User group 1 and the home area code tables must be programmed before using the Fax-On-Demand feature. A fax library must also be established. See the "Fax-On-Demand" on **page 950**.
	Incorrect switch or jumper settings on the Fax Card.	Ensure that the switches and jumpers are set in the correct positions.
	Defective Fax Card	Replace the Fax Card.
	Improper drivers are loaded	Load the correct drivers. See if the fax ports are listed in the summary.txt (in the Avdap folder).
Unable to record any voice mail greetings or messages and/or unable to hear any voice prompts	Loose/defective cabling between the external voice processor and the chassis	Check the cabling.
Japanese language voice prompts unavailable	Japanese voice prompts not installed	Install the Japanese voice prompts as outline in the Installation chapter in the *MiVoice Office 250 Installation Manual* .
	User and/or programming error	See "Multilingual Capability" on **page 314**.

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Chapter 17

VOICE PROCESSOR REPORTS

INTRODUCTION

With v5.0, the DB Programming Report Programming function has been moved to the Mitel System Administration & Diagnostics application, enhanced with a new look and feel, and renamed as Database Reporting. Previous reports programmed in DB Programming are discarded and must be newly programmed using Database Reporting in System Administration & Diagnostics. For details about Database Reporting, refer to the *Mitel System Administration & Diagnostics Help*.

REPORT PARAMETERS

The voice processor can print reports from the external voice processing system to a printer or to a file for storage. The reports include Application and Channel Statistics, Directory Listings (by last name or first name or extension), Group List reporting, Fax Document Usage, Fax Delivery, and Directory Sort Order. The Report Parameters folders and fields display with a red "X" when using UVM because with UVM, these reports cannot be retrieved.

REPORT OPTIONS

The reports include the following:

APPLICATION AND CHANNEL STATISTICS

The following information appears individually for each application and as a summary for all applications.

- **Description and extension number of the application:** Shows the programmed name for the application. (Call Routing Announcement applications are all listed together by extension.) The applications are listed in the following order: Message Notification/Retrieval, Voice Mail, Auto Attendant, and Call Routing Announcement. (Auto Attendant Recall applications are reported within the Auto Attendant information.)

- **Incoming calls:** Shows the total number of calls received by that extension number. This is shown as a combined total for Call Routing Announcement applications.

- **Outgoing calls:** Shows the Message Notification/Retrieval application. These are the remote message notification calls placed by the Voice Mail application.

- **Connect minutes:** Shows the total time spent on incoming and outgoing calls (if any) combined. This is shown as a combined total for Call Routing Announcement applications.

- **Minutes per call:** Shows a combined total of the average amount of time spent on each call in minutes and seconds for Call Routing Announcement applications.

- **Transfers to Operator:** Shows the number of times a caller (within Voice Mail or Auto Attendant) presses the dial pad button 0 for operator access.

- **Voice Mail messages left:** Appears in the summary section only. It shows how many voice mail messages were left in all mailboxes combined.

- **Channel statistics:** Includes activity data of all applications. It shows, in 30-minute segments, the total number of minutes and seconds that all of the voice processing voice channels were busy simultaneously. The detailed segments begin at 07:00 AM and conclude at 06:00 PM. The "Off Peak Hours" segment shows statistics for the remaining time period (6:00 PM to 7:00 AM). This section ends with a grand total of busy channel occurrences for each of the days being reported.

  If the number of voice channels programmed in DB Programming is greater than the number of actual channels available, the statistics reported will be based on the number available and ***not*** the programmed number.

DIRECTORY LISTING REPORTS

You can sort directory listings by first name, last name, or extension/mailbox number (directory number). The listings show the mailbox description or extension ID, the mailbox/extension number, the message notification phone for mailboxes, and mailbox information. The mailbox information shows whether the mailbox is marked Private and/or Unlisted. (An X appears in the Mailbox field to indicate a mailbox that is neither Private nor Unlisted and a blank indicates that it is an extension ID.)

GROUP LIST REPORT

The Group List report provides a printed copy of the system group lists. The report identifies the group list number, the list description, and the mailboxes included in the group list.

FAX DOCUMENT USAGE REPORT

This report lists all documents in the fax library by document number. Each entry shows the document number, description, how many times it was delivered to callers, the last request date, and the last revision date and time. If the document has not been revised, it shows the import date and time.

FAX DELIVERY REPORTS

The Fax Delivery Report includes information for up to 200 fax delivery attempts. Each entry contains the date and time of the delivery attempt, the date and time the fax was requested, the delivery status (Successful, Busy, Call Failed, or Transmission Error), the fax number, and the list of requested documents. The Fax Delivery Report displays an asterisk (*) immediately to the left of the delivery status in the Fax Delivery Report for an entry representing a fax delivery that failed and was removed from the delivery queue. Fax deliveries can fail for many reasons, but the most common problem is that the fax number entered was not a fax machine, but was a company's main number or answering service. Review the Fax Delivery Report on a regular basis to check for delivery failures.

DIRECTORY SORT ORDER

(Available only if Directory Listing Reports are selected.) You can use the Directory Sort Order lists to select a sorting order from the following options: First Name, Last Name, or Directory Number.

USING AUTOMATIC REPORT GENERATION

You can use Automatic Report Generation to:

- Set the day and time the system for automatic report printouts.

- Enable automatic report printouts.

- Clear system statistics.

- List the reports to print.

- Sort the Directory Listing report.

To enable automatic report generation:

1. Select Voice Processing – Report Parameters – **Automatic Report Generation**.

2. Select **Enable**.

3. In the **Value** column, select the check box to select the option. The field changes to **Yes**. To deselect the option, clear the check box.

4. Click out of the field or press **ENTER** to save the change.

REPORT SELECTION

You can select the report types that you want to generate.

To select report types:

1. Select Voice Processing – Report Parameters – **Automatic Report Generation**.

2. Double-click **Report Generation**. The report types appear in the right pane.

3. In the **Value** column, select the check box. The field changes to **Yes**. To deselect the option, clear the check box.

4. Click out of the field or press **ENTER** to save the change.

STATISTICS TO CLEAR

You can select the following statistics to be cleared after a report is printed:

*   **Application and channel**: Clears the information that is listed in the Application and-channel Statistics report only.

*   **Fax Delivery and Document Usage**: Clears all statistics for the Fax Delivery report and the request count and last request date for the Fax Document Usage report.

*   **Mailbox**: Clears all mailbox statistics.

To mark a statistic to be cleared:

1. Select Voice Processing – Report Parameters – **Automatic Report Generation**.

2. Select **Statistics to clear**.

3. In the **Value** column, select the check box. The field changes to **Yes**. To deselect the option, clear the check box.

4. Click out of the field or press **ENTER** to save the change.

PRINT DAY

You can select the day in which the system prints the automatic reports.

To select the day to print automatic reports:

1. Select Voice Processing – Report Parameters – **Automatic Report Generation**.

2. Select **Print Day**.

3. In the **Value** column, select the day from the list.

4. Click out of the field or press **ENTER** to save the change.

PRINT TIME

You can select the time of day in which the system prints the automatic reports.

To select the time of day for printing automatic reports:

1. Select Voice Processing – Report Parameters – **Automatic Report Generation**.

2. Select **Print Time**.

3. In the **Value** column, type or select the time (AM or PM).

4. Click out of the field or press **ENTER** to save the change.

USING MANUAL REPORT GENERATION

Manual Report Generation lists report types that you can print and the option to sort the Directory Listing report. You can also clear statistics after printing the reports.

REPORT SELECTION

Double-click **Report Selection** to view the list of available report types.

To enable a report:

1. Select Voice Processing – Report Parameters – **Manual Report Generation**.

2. Double-click **Report Generation**. The report types appear in the right pane.

3. In the **Value** column, select the check box. The field changes to **Yes**. To deselect the option, clear the check box.

4. Click out of the field or press **ENTER** to save the change.

STATISTICS TO CLEAR

You can select the following statistics to be cleared after a report is printed. See **page 1027** for descriptions.

To mark a statistic to be cleared:

1. Select Voice Processing – Report Parameters – **Manual Report Generation**.

2. Select **Statistics to clear**.

3. In the **Value** column, select the check box. The field changes to **Yes**. To deselect the option, clear the check box.

4. Click out of the field or press **ENTER** to save the change.

PRINTING REPORTS

To print the selected reports:

1. Select Voice Processing – Report Parameters – **Manual Report Generation**.

2. Right-click **Report Selection**, and then select **Print Selected Reports**.

3. From the list, select the output device. For a UVM system, the options are FILE or LPT1. For an external voice processing system, the options are FILE or EMAIL.

    *   *If "LTP1" is selected,* a printer must be attached to the parallel port on the voice mail server.

    *   *If "File" is selected,* you must program the Output Destination field.

4. Click out of the field or press **ENTER** to save the change.

The following applies only if output device is "FILE" or "E-mail."

*   *If the Output Device is File*,

    a. In the **File** box, type the name of the file (for example, C:\avdap\reports.txt).

    b. Click out of the field or press **ENTER** to save the change. The system validates that the specified drive output is defined, but does ***not*** validate that the path exists on the voice processing computer. Instead, the database manager verifies that the file syntax is correct (for example, check for valid characters). If only the file name is entered, the system saves the report using the file name under the Avdap directory.

*   *If the Output Device is Email,*

    a. Type the e-mail address (for example, johndoe@abc.com) in the box.

    b. Click out of the field or press **ENTER** to save the change.

*   *If the e-mail address is invalid or if the destination mailbox is full*, the system generates a voice mail message indicating that the report could not be delivered. This message is left in the designated system administrator voice mailbox.

Chapter 18

SYSTEM DIAGNOSTICS

INTRODUCTION

System diagnostics are provided to assist trained personnel in monitoring and maintaining the functional health of the system. This chapter provides fundamental instruction for interpreting the output data from the utilities.

Depending on the problem and the data that has been collected during troubleshooting, technical support personnel may require additional information to perform their analysis. In many instances, diagnostic utility information and instruction is provided by the product specialist and may not be covered in this chapter. Some diagnostics should be performed only when directed by authorized technical support personnel.

This section provides information about diagnostic applications available throughout the system and associated software components. In the following sections, the applications are organized and grouped by their location in the system or a component within the system. The information that is provided for each application includes:

- Where the application is located in the system.

- Instructions on how to implement the application.

- The purpose of the application; for example, identification of the information captured.

- How to use the information from the application for troubleshooting.

DIGITAL TRUNK DIAGNOSTICS

The Digital Trunk Diagnostics option in the Diagnostics Management Utility (DMU) provides enhanced diagnostic capability for digital trunks. Through the DMU interface, Digital Trunk Diagnostics provides data about activity occurring on a system's trunks attached to the single-port T1/E1/PRI Module (T1M), the Dual T1/E1/PRI Module (T1M-2), and the Basic Rate Module (BRM-2) that supports the S/T interface of Integrated Services Digital Network (ISDN) trunks.

The following characteristics apply to the DMU Digital Trunk Diagnostics screens:

- **Maximum of five diagnostic connections per module**: Each module can support up to five simultaneous diagnostic sessions. The five-session limit prevents the system from overloading.

- **Self-refreshing screens**: With the exception of the following screens, all screens refresh in real-time:
    - Menu screens
    - The Module Selection Rejection screen
    - Version screens

- **Detecting installed equipment**: The DMU verifies whether the system is equipped with appropriate hardware.

This section contains the following information:

- **Navigating Through the Digital Trunk Diagnostics Screens** below
- DMU Main Screen on **page 18-1034**
- Module Selection Screen on **page 18-1034**
- Dual T1/E1/PRI Module (T1M-2) Diagnostics Screens on **page 18-1036**
- T1/E1/PRI Module (T1M) Diagnostics Screens on **page 18-1066**
- Basic Rate Module (BRM-2) Diagnostics Screens on **page 18-1089**

NAVIGATING THROUGH THE DIGITAL TRUNK DIAGNOSTICS SCREENS

Navigation controls for the Digital Trunk Diagnostics screens work in the following manner:

- For the menu screens use the UP ARROW or the DOWN ARROW to move between menu items.

- A menu item may also be activated by using a shortcut key. Shortcut keys are the letters preceding the menu items, and either a capital or lower case letter activates the selection.

- To activate a menu item, highlight it and press **Enter**.

- Navigation of the menu items follows a looping path. That is, pressing the DOWN ARROW from the last menu item highlights the first menu item; and, pressing the UP ARROW from the first menu item highlights the last menu item.

Additional navigational or command instructions are displayed on individual screens.

DMU MAIN SCREEN

The DMU Main Menu screen shows the Digital Trunk Diagnostics option, as shown in **Figure 3**.

```
             DIAGNOSTICS MANAGEMENT UTILITY

               DATE = May 23, 2007 11:23:58
               DMU VERSION = 1.0     PID = 3142
               Executing Locally 172.16.9.72
          INTER-TEL CONFIDENTIAL AND PROPRIETARY


        +---------------------------------------+
        :                DMU Main Menu          :
        +---------------------------------------+
        :* Command Line              Help       :
        :  Configuration             Exit DMU   :
        :  Digital Trunk Diagnostics            :
        :                                       :
        :                                       :
        :                                       :
        :                                       :
        +---------------------------------------+



Use Up and Down Arrow keys to Navigate
Press Enter to Make Selection
```

MODULE SELECTION SCREEN

Selecting the Digital Trunk Diagnostics option displays the Module Selection screen, as shown in **Figure 4**. From the Module Selection screen you select the bay of the installed digital trunk module (T1M, T1M-2, or BRM-2) you want to analyze. This screen allows you to select the Digital Trunk module (T1M, T1M-2, or BRM-2) and connect to its DMU diagnostics screens.

```
         Digital Trunk Diagnostics
         =========================
         A. Bay 1 - N/A [empty]
         B. Bay 2 - N/A [LSM4]
         C. Bay 3 - T1M-2
         X. Exit
         Enter selection: █



         System Reference Clock: Internal (Motherboard)
```

When this screen appears, the first valid selection is highlighted. The module selection screen for Digital Trunk Diagnostics shows the elements described in the following paragraphs.

Module names for each Bay include:

- N/A [empty]

- N/A [LSM2]

- N/A [LSM4]

- N/A [SLM4]

- T1M

- T1M-2

- BRM-2

Pressing keys provides shortcuts to module selections. **Table 168** summarizes valid keyboard shortcuts.

Table 168:   Keyboard shortcuts for Module Selection Screen

KEY.	SCREEN ELEMENT
A or a	Bay 1
B or b	Bay 2
C or c	Bay 3
X or x	Exit

The screen displays the source used as the System Reference Clock. Valid sources include:

- Internal (Motherboard)

- Bay x - T1M

- Bay x - T1M-2 Port *<n>*

- Bay x - BRM-2

Connecting to module diagnostics may be rejected by presenting the Module Selection Rejection screen if the maximum number of five connections has already been established with that module. See **Figure 5**.

```
Sorry.  The maximum number of connections to the selected module has
been exceeded.

Press any key to exit
```

NOTE	Pressing any key returns control back to the Module Selection screen.

DUAL T1/E1/PRI MODULE (T1M-2) DIAGNOSTICS SCREENS

The following sections describe the DMU screens that support Dual T1/E1/PRI Module diagnostics. This screen periodically refreshes options D and E.

MAIN SCREEN

This section describes the main screen for Dual T1/E1/PRI Module (T1M-2) diagnostics.

To display the T1M-2 diagnostics main screen:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a Bay containing a T1M-2 module. The main screen appears, as shown in **Figure 6**.

```
                                                              [Bay 1]


        Dual T1/E1/PRI Module
        =====================
        A. Status
        B. Busy Out
        C. Version
        D. Port 1 [T1]
        E. Port 2 [T1/PRI]
        F. DSP Audio Diagnostics
        G. Clock Frequency
        H. Timers
        <. Back
        X. Exit
        Enter selection: █
```

The **Status** selection is highlighted upon entry to this screen.

Port types appearing in brackets include:

- T1

- T1/PRI

- E1/PRI

- UNEQUIPPED

STATUS

The following section describes the Status screen for the T1M-2 module diagnostics. This screen refreshes periodically.

To display the Status screen for T1M-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M-2 module.

3. Select **Status**. The STATUS screen appears, as shown in **Figure 7**.

```
[Bay 1]                      << S T A T U S >>            's' - SYNC Status screen
                                                          '.' - To exit screen


         | Port Type        SYNC Status            REF LED         BUSY LED
Port 1 | T1                 INIT-TRBL/RED-ALARM     OFF             OFF
Port 2 | E1/PRI             SYNC                    ON              ON

Channel        |                      1 1 1 1 1 1 1 1 1 1 2 2 2 2 2
Number         | 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4
Port 1         | + + + + + + + + + + + + + + + + + + + + + + + +
 Span-Side EC  |
 System-Side EC |

Channel        |                      1 1 1 1 1 1 1 1 1 1 2 2 2 2 2 2 2 2 2 2 3
Number         | 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0
Port 2         | - - B I I 0 0 0 + + + + + + + + + + + + + + B + + - - - - - -
 Span-Side EC  |      *                                        *
 System-Side EC |      *                                        *

Legends: B - Busied out                    + - Idle
         i - Busying out incoming call      X - Offline
         o - Busying out outgoing call      - - Unequipped
         I - Incoming call
         0 - Outgoing call                  * - Echo Canceller active
```

The Status screen for the T1M-2 module diagnostics shows the following elements:

The **Port Type** field displays one of the following states:

- T1
- T1/PRI
- E1/PRI
- UNEQUIPPED

The **SYNC Status** field displays one of the following values:

- RED-ALARM
- INIT-TRBL
- INIT-TRBL/RED-ALARM
- SYNC
- Blank, if the port is unequipped

NOTE | The RED-ALARM, INIT-TRBL, INIT-TRBL/RED-ALARM values blinks to gain the user's attention.

The **REF LED** and **BUSY LED** fields display a value of OFF, ON, or blank if the port is unequipped.

Channel Number rows count to the maximum number of channels for the corresponding port type:

- 23 for T1/PRI

- 24 for T1

- 30 for E1/PRI

In addition to the general navigation guidelines described in "Navigating Through the Digital Trunk Diagnostics Screens" on **page 18-1033**, the following controls apply:

Pressing the S key or the s key transfers control to the SYNC Status screen.

STATUS – SYNC STATUS

The following section describes the SYNC STATUS screen for T1M-2 modules. The SYNC STATUS screen shows the same information that the M8000 and M8001 System Message Print messages show but in a real-time format instead of a log format. For descriptions of the various Initial-Trouble/Red-Alarm conditions, refer to the *Message Print Diagnostics Manual*, part no. 550.8018. This screen refreshes periodically.

To display the SYNC STATUS screen for T1M-2 diagnostics:

1. From the DMU Main Menu, select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M-2 module.

3. Select **Status**.

4. While in the Status screen, press the S key or the s key. The SYNC STATUS screen for T1M-2 (or T1M) appears, as shown in **Figure 8**.

```
[Bay 1]              << S Y N C   S T A T U S >>       's' - Status screen
                                                        '.' - To exit screen


Port 1 (T1):                                  | Initial | Red    |
    Condition                                 | Trouble | Alarm  |
    Receiver Loss of Signal from Network..... |    *    |   *    |
    Transmitter Loss of Signal.............   |    *    |        |
    Receiving Yellow Alarm.................   |         |   *    |
    Receiving Blue Alarm...................   |         |        |
    T1 Signal Out-of-Frame.................   |         |        |
    Severely Errored Seconds...............   |    *    |        |
    Transmitting Blue Alarm................   |         |   *    |
    Remote Loopback State..................   |         |        |

Port 2 (E1/PRI):                              | Initial | Red    |
    Condition                                 | Trouble | Alarm  |
    Receiver Loss of Signal from Network..... |    *    |   *    |
    Transmitter Loss of Signal.............   |    *    |        |
    Loss of Frame Synchronization..........   |         |        |
    Loss of Multiframe Synchronization.....   |         |        |
    Alarm Indication Signal (AIS) Reception. |         |        |
    Remote Alarm Indication Signal (RAI) Reception |    |        |
    Transmitting AIS.......................   |         |        |
```

The Sync Status screen for T1M-2 module diagnostics shows the following elements:

• The heading for each port displays the port type in parenthesis. Valid port types include:
  • T1
  • T1/PRI
  • E1/PRI
  • UNEQUIPPED

• The single-port T1M does not indicate a Port number.

- If a port is unequipped, that port does not show a corresponding SYNC status table.

- The tables for T1 and T1/PRI are different from the table for E1/PRI.

In addition to the general navigation guidelines described in "Navigating Through the Digital Trunk Diagnostics Screens" on , the following control applies:

Pressing the S key or the s key transfers control to the Status screen.

BUSY OUT

The following section describes the Busy Out screen for the T1M-2 module diagnostics. This screen refreshes periodically.

To display the Busy Out screen for T1M-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M-2 module.

3. Select **Busy Out**. The BUSY-OUT screen appears, as shown in **Figure 9**.

```
Bay 1]                    << B U S Y - O U T >>        't'/'u' - Tag/Untag
                                                       'e' - EC Status
                                                       Arrow keys - Change channel
                                                       '.' - To exit screen


Port 1
  Channel     |                    1 1 1 1 1 1 1 1 1 1 2 2 2 2 2
  Number      | 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4
  Status      | + + + + + + + + + + + + + + + - - - - - - - - -
  Busy-out Tag |

Port 2
  Channel     |                    1 1 1 1 1 1 1 1 1 1 2 2 2 2 2 2 2 2 2 2 3
  Number      | 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0
  Status      | + + B + + I + + + + + o + + + + + + + + + I I I I I O O
  Busy-out Tag |     * *     *               *

  Legends: B - Busied out                  + - Idle
           i - Busying out incoming call   X - Offline
           o - Busying out outgoing call   - - Unequipped
           I - Incoming call
           O - Outgoing call               * - Tagged for busy out
```

The Busy Out screen for T1M-2 module diagnostics shows the following elements:

- **Port 1** is highlighted and flashing upon entry to this screen.

- The **Channel Number** identifies the channels available with the trunk programmed for the module port.

- The **Status** line identifies the status of the channel above it.

- The **Busy-out Tag** line identifies the Busy Out condition of the channel above it.

- The **Legends** area defines the Status line and Busy-out Tag line symbols.

Instead of the navigation guidelines described in "Navigating Through the Digital Trunk Diagnostics Screens" on **page 18-1033**, the following controls apply:

- Pressing the UP ARROW or DOWN ARROW switches the selection between ports.
- From a port selection, pressing the LEFT ARROW moves the cursor to the last channel in that port.
- From a port selection, pressing the RIGHT ARROW moves the cursor to the first channel in that port.
- Pressing the LEFT ARROW or RIGHT ARROW moves the cursor between channels in the same port.
- Pressing the LEFT ARROW key from the first channel moves the cursor to the last channel in the same port.
- Pressing the RIGHT ARROW from the last channel moves the cursor to the first channel in the same port.
- Pressing the UP ARROW while a channel is selected selects the channel's port.
- Pressing the DOWN ARROW while a channel is selected select the other port.
- If a port is selected, pressing the T key or the t key tags all channels in that port for busy-out.
- If a port is selected, pressing the U key or the u key untags all channels in that port for busy-out.
- If a channel is selected, pressing the T key or the t key tags that channel for busy-out
- If a channel is selected, pressing 'the U key or the u key untags that channel for busy-out

VERSION INFORMATION

The following section describes the Version Information screen for T1M-2 module diagnostics.

To display the Version Information screen for T1M-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M-2 module.

3. Select **Version**. The Version Information screen appears, as shown in **Figure 10**.

```
Version Information                                                    [Bay 1]
===================

Hardware
  Version:            0x2

FPGA
  Version:            0x2

T1 Application
  Version:            2.3.28_asserts
  Build Date:         Wed May 23 2007 11:23:58
  Build directory:    /home/ymc/workspace/RemoteOffice/intl/apps/dtl

Firmware
  Version:            2.3.ymc_asserts
  Build Date:         Wed May 23 2007 11:23:58
  Build directory:    /home/ymc/workspace/RemoteOffice/intl/firmware/dtl

DSP
  Version:            01.05.12
  Build Date:         Mon Dec 18 10:36:27 2006
```

The format of the version numbers varies in accordance with data that is available for each software/hardware module.

PORT 1 OR 2

The following section describes the Port screen for T1M-2 module diagnostics.

To display the Port screen for T1M-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M-2 module.

3. Select **Port**. The Port screen appears, as shown in **Figure 11**.

```
                                                                      [Bay 1]



        Port 1 [T1/PRI]
        ===============
        A. Layer 1
        B. Channels
        C. T1 Performance
        D. Call Statistics
        E. ISDN Diagnostics
        F. Loopback
        G. Echo Canceller Channels
        <. Back
        X. Exit
        Enter selection: █

```

The **Layer 1** option is highlighted upon entry to this screen.

The screen title displays the corresponding port number and the port type in brackets. Port types include:

- T1

- T1/PRI

- E1/PRI. Note that the port number is not displayed for a T1M.

ISDN Diagnostics appears only when the port is programmed T1/PRI or E1/PRI trunk.

The **Performance** option changes to conform with the port type. That is, when the port is programmed as T1 or T1/PRI, the option appears as **T1 Performance**, and when the port is programmed as E1/PRI, the option appears as **E1 Performance**.

PORT 1 OR 2 – LAYER 1

The following section describes the Open Systems Interconnect (OSI) model Layer 1 screen for T1M-2 module diagnostics. This screen is useful to get a glance at a T1/E1/PRI port Layer-1 status and the channel activity on that port, all in one screen. This screen refreshes periodically.

To display the LAYER-1 screen for T1M-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M-2 module.

3. Select **Port 1** or **Port 2**.

4. Select **Layer 1**. The LAYER-1 screen for the port appears, as shown in **Figure 12**.

```
[Bay 1] [Port 1]          << L A Y E R - 1 >>        'A' - Toggle Alarm Logging
                                                     '.' - To exit screen

Span Type............ E1-PRI                         Red Alarm..... No
Card Slot............ 1           TCR Local Loopback.. No   Frame Errors.. 0
Com. Chan. Signalling Yes         TCR Remote Loopback. No   Slips......... 0
CRC Enabled.......... Yes         FMR Remote Loopback. No   Sync.......... Yes
HDB3 Enabled......... Yes         FMR Local Loopback.. No   MF Sync....... Yes
Term. Resistor Jumper 120 ohms    TX Loss of Signal... No   CRC Sync...... Yes
TX Pulse Template.... 120 ohms    TX RAI.............. Auto  Si2........... 1
Clock Timing......... Slave       TX AIS.............. No    Si1........... 1
Ref. Clock Source.... ANCC        RX Loss of Signal... No    FEBEs......... 0
System Sync Source... This E1     RX RAI.............. No     CRC Errors.... 0
Local Sync Request... Yes         RX AIS.............. No
Clock Mode........... CEPT        Network Loopback.... N/A   Alarm Logging. Yes

B-CHANNEL|              1 1 1 1 1 1 1 1 1 1 2 2 2 2 2 2 2 2 2 2 3
NUMBER   | 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0
STATUS   |
```

This screen displays a port number field at the top of the screen.

PORT 1 OR 2 – CHANNEL SELECTION

The following section describes the port Channel Selection screen for T1M-2 module diagnostics. When diagnosing a T1M-2 module, the Channel Selection screen presents underlined port-number fields.

To display the Channel Selection screen for T1M-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M-2 module.

3. Select **Port**.

4. Select **Channels**. The Channel Selection screen for the selected port appears, as shown in **Figure 13**.

```
                                                              [Bay 1]
        Channel Selection [Port 1]
        ===========================
        Channel #1        Channel #16
        Channel #2        Channel #17
        Channel #3        Channel #18
        Channel #4        Channel #19
        Channel #5        Channel #20
        Channel #6        Channel #21
        Channel #7        Channel #22
        Channel #8        Channel #23
        Channel #9        Channel #24
        Channel #10       Channel #25
        Channel #11       Channel #26
        Channel #12       Channel #27
        Channel #13       Channel #28
        Channel #14       Channel #29
        Channel #15       Channel #30

        <. Back
        X. Exit
        Enter selection: █
```

The **Channel #1** selection is highlighted upon entry to this screen.

The number of channels displayed varies according to the port type, as follows:

- T1 = 24 channels

- T1/PR I = 23 channels

- E1/PRI = 30 channels

After using the ARROW keys to select the channel you want, press **Enter** to go to the selected Channel screen.

PORT 1 OR 2 – CHANNEL SELECTION – CHANNEL

The following section describes the port Channel screen on T1M-2 module diagnostics. This screen refreshes periodically.

To display a port Channel screen for T1M-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M-2 module.

3. Select **Port**.

4. Select **Channels**.

5. Select the channel you want to diagnose. The Channel screen for the selected port appears, as shown in **Figure 14**.

```
[Bay 1] [Port 1] Channel #1                   Arrow keys - Move
                                              'c' - Toggle Diag. Logging
CO Line Type:          Unknown                '.' - To exit screen
Current State:         Non-Functional
Current Substate:      Non-Functional Entry
Next Substate:         Global Relay Delay
Response Buffer:       $62 $1A $01 $3C $00 $0A
Static Line Type:      %00000000
Dynamic Line Type:     %00000000
Last Digit Dialed:     $00
TX Gain:               %00000000
RX Gain:               %00000000
Clear Channel:         No
Miscellaneous Flags:   %00000000
Rockwell Address:      $02
T1 Channel:            $01
RX Signalling:         A=1 B=1
TX Signalling:         A=0 B=0
Channel LED:                        CO Task Diagnostics Logging:  Yes
```

For non-ISDN channels, a **CO Task Diagnostics Logging** field appears in the lower right corner. This field can be switched between Yes and No by pressing the C key or the c key. The output controlled by this field goes into the T1/DT1/BRI Diagnostics log.

PORT 1 OR 2 – T1 PERFORMANCE

The following section describes the port T1 PERFORMANCE screen supporting T1M-2 module diagnostics. This screen is useful to view the same Hourly and Daily error counts that are available through the DB Programming. This screen refreshes periodically.

To display a port T1 PERFORMANCE screen for T1M-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M-2 module.

3. Select **Port 1** or **Port 2**, as applicable.

4. Select **T1 Performance**. The T1 PERFORMANCE screen for the selected port appears, as shown in **Figure 15**.

```
11:23:58 Wednesday, May 23, 2007                              [Bay 1]

                    << T 1   P E R F O R M A N C E >>

Port #1
                       Current Counts          Error Thresholds
T1 Performance Parameter   Hourly    Daily       Hourly      Daily
Bipolar Violations              0     1436        53360     133400
CRC Errors                      0       13        53184     132960
Out-of-Frame Errors             0       91            7         17
Frame Slip Errors              91    61733            2          4
Errored Seconds                 0        4          259        648
Severely Errored Seconds        0        0           40        100
Unavailable Seconds             0        0           10         10




Press any key to exit
```

This screen shows only the counts for the current hour and current day, as opposed to the DB Programming data, which shows a history of 24 hourly counts and a history of seven daily counts for each T1 Performance parameter.

PORT 1 OR 2 – CALL STATISTICS

The following section describes the port CALL STATISTICS screen supporting T1M-2 module diagnostics. This screen refreshes periodically.

To display a port CALL STATISTICS screen for T1M-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M-2 module.

3. Select **Port 1** or **Port 2**, as applicable.

4. Select **Call Statistics**. The CALL STATISTICS screen for the selected port appears, as shown in **Figure 16**.

```
[Bay 1] [Port 1]   << C A L L   S T A T I S T I C S >>   'c' - To clear counts
                                                         '.' - To exit screen
Current Time:        11:23:58 Wednesday, May 23, 2007
Last Time Cleared:   11:23:58 Wednesday, May 23, 2007

           Incoming   Outgoing                    Incoming   Outgoing
Channel      Calls      Calls        Channel        Calls      Calls
   1           0          0             13            0          0
   2           0          0             14            0          0
   3           0          0             15            0          0
   4           0          0             16            0          0
   5           0          0             17            0          0
   6           0          0             18            0          0
   7           0          0             19            0          0
   8           0          0             20            0          0
   9           0          0             21            0          0
  10           0          0             22            0          0
  11           0          0             23            0          0
  12           0          0

   Total # of Connected Calls:   0
```

The call counts are reset to zero when the module powers up; therefore the counts are not preserved across module/system reset.

This screen displays 23 channels for T1/PRI, 24 channels for T1, or 30 channels for E1/PRI ports.

PORT 1 OR 2 – ISDN DIAGNOSTICS

The following section describes the port ISDN Diagnostics screen supporting T1M-2 module diagnostics.

To display a port Call Statistics screen for T1M-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M-2 module.

3. Select **Port 1** or **Port 2**, as applicable.

4. Select **ISDN Diagnostics**. The ISDN Diagnostics screen for the selected port appears, as shown in **Figure 17**.

```
                                                              [Bay 1]


        ISDN Diagnostics [Port 1]
        =========================
        A. Data Link Status
        B. D-Channel Diagnostics
        <. Back
        X. Exit
        Enter selection: █
```

The **Data Link Status** selection is highlighted upon entry to this screen.

Typing the selection shortcuts—A or a, B or b, <, X or x— activates the corresponding selection.

Pressing **Enter** activates the highlighted selection.

The cursor is located at the **Enter selection** prompt upon entry to this screen. Once the UP ARROW or DOWN ARROW has been pressed, from that time forward, the cursor is located at the end of the highlighted selection.

PORT 1 OR 2 – ISDN DIAGNOSTICS – DATA LINK STATUS

The following section describes the port ISDN Data Link Status screen supporting T1M-2 module diagnostics. This screen is useful to troubleshoot OSI model Layer 2 problems on an ISDN connection. The screen refreshes periodically.

To display a port ISDN Data Link Status screen for T1M-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M-2 module.

3. Select **Port 1** or **Port 2**, as applicable.

4. Select **ISDN Diagnostics**.

5. Select **Data Link Status**. The ISDN Data Link Status screen for the module port appears, as shown in **Figure 18**.

```
[Bay 1] Port #1                                '.' - To exit screen

PRI (ISDN) Configuration
Switch Type... DMS-100          K.......... 7
T200......... 1000 ms           N200....... 3
T203......... 10000 ms          N201....... 260
T204......... 2000 ms           N204....... 3

Layer-2 Network Mode:  DISABLED


Data Link Control Block
D-Channel Status... UP
Sapi.............. $00
Tei............... $00
State............. MULTIPLE FRAME ESTABLISHED
VA................ $37
VS................ $37
VR................ $3B

L2_Queue_Usage.... $00
L2_Queue_Usage_Max $00
```

The **Any other key - Next Port** option in the upper right corner of the screen only if two ports are equipped. If the other port on a T1M-2 is unequipped, the option is omitted. The option may appear or disappear if the other port is equipped or unequipped while on this screen.

PORT 1 OR 2 – ISDN DIAGNOSTICS – D-CHANNEL DIAGNOSTICS

The following section describes the port D-Channel Diagnostics screen supporting T1M-2 module diagnostics. The D-Channel Diagnostics screen allows you to enable or disable the D-Channel Diagnostics for the current port. This screen refreshes periodically.

To display a port ISDN Data Link Status screen for T1M-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M-2 module.

3. Select **Port 1** or **Port 2**, as applicable.

4. Select **ISDN Diagnostics**.

5. Select **D-Channel Diagnostics**. The D-Channel Diagnostics screen for the module port appears, as shown in **Figure 19**.

```
                                                              [Bay 1]


        D-Channel Diagnostics: Port 1: [DISABLED]
        =========================================
        A.  Enable D-Channel Diagnostics (Level-1)
        B.  Enable D-Channel Diagnostics (Level-2)
        C.  Disable D-Channel Diagnostics
        D.  Display D-Channel Diagnostics
        <.  Back
        X.  Exit
        Enter selection:
```

The **Enable D-Channel Diagnostics (Level-1)** selection is highlighted upon entry to this screen.

For the T1M no reference to Port # appears in the menu title.

The state of the current port's D-Channel Diagnostics is enclosed in brackets and can have one the following values:

- DISABLED

- LEVEL-1

- LEVEL-2

- UNDEFINED

The **Display D-Channel Diagnostics** option has no effect if the D-Channel Diagnostics is disabled.

PORT 1 OR 2 – ISDN DIAGNOSTICS – D-CHANNEL DIAGNOSTICS – DISPLAY

The following section describes the port D-Channel Diagnostics Display screen supporting T1M-2 module diagnostics. The D-Channel Diagnostics Display screen allows you to view the D-Channel Diagnostics in real-time.

To display a port D-Channel Diagnostics Display screen for T1M-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M-2 module.

3. Select **Port**.

4. Select **Port 1** or **Port 2**, as applicable.

5. Select **ISDN Diagnostics**.

6. Select **D-Channel Diagnostics**.

7. Select **Enable D-Channel Diagnostics Level-1** or **Level-2**, as applicable.

8. Select **Display D-Channel Diagnostics**, The D-Channel Diagnostics Display screen for the module port appears, as shown in **Figure 20**.

```
                                                                    [Bay 1]

D-Channel Diagnostics: Port 1: [LEVEL-1]
=========================================

Pressing any key now will start the D-channel Diagnostics real-time display on
the screen.  Once the D-Channel Diagnostics real-time display has started,
pressing the '.' key will exit back to D-Channel Diagnostics menu.

Press any key to continue: █




```

The state of the D-Channel diagnostics for the current port is displayed in brackets. Possible values include:

- LEVEL-1

- LEVEL-2

- UNDEFINED

PORT 1 OR 2 – LOOPBACK

The following section describes the port Loopback screen supporting T1M-2 module diagnostics. The LOOPBACK screen allows you to put the current port into one of four loopback modes and to transmit and detect specific bit patterns. This screen refreshes periodically.

To display a port Loopback screen for T1M-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M-2 module.

3. Select **Port**.

4. Select **Port 1** or **Port 2**, as applicable.

5. Select **Loopback**. The LOOPBACK screen for the module port appears, as shown in **Figure 21**.

```
[Bay 1] [Port #1]        << L O O P B A C K >>        SPACE - Activate Selection
                                                      Arrow keys - Change Selection
                                                      '.' - To exit screen


Loopback Type         | Loopback Status
Digital Local         | DISABLED
Digital Remote        | DISABLED
Transceiver Local     | DISABLED
Transceiver Remote    | ENABLED (remotely)



Bit Pattern           | Pattern Transmit | Pattern Received
Loop-up               | DISABLED         | NO
Loop-down             | DISABLED         | NO
Framed 0101           | DISABLED         | ---
```

The Loopback screen for T1M-2 module diagnostics shows the following elements:

- The **Digital Local** loopback is selected upon entry to this screen.

- The **Loopback Status** and **Pattern Transmit** fields display either DISABLED or flashing ENABLED. Additionally, the Transceiver Remote loopback type may also display a flashing ENABLED (remotely) value for its Loopback Status.

- The **Pattern Received** fields display either YES or NO, except for the Framed 0101 bit pattern which always appears as **---**.

- The **Bit Errors** field applies to the Framed 2^XX-1 bit patterns only and displays a number only when the corresponding bit pattern is active. The other bit patterns always display as **---**.

The following controls apply:

- Pressing the UP ARROW or the LEFT ARROW moves the highlight to the next Loopback Type field or Bit Pattern field in the upward direction.

- Pressing the DOWN ARROW or the RIGHT ARROW moves the highlight to the next Loopback Type field or Bit Pattern field in the downward direction.

- Pressing the SPACEBAR activates the highlighted Loopback Type or Bit Pattern. When a Loopback Type or Bit Pattern is enabled, all the other Loopback and Bit Patterns are disabled and the corresponding Loopback Status or Pattern Transmit entry is enabled.

PORT 1 OR 2 – ECHO CANCELLER SELECTION

The following section describes the port Echo Canceller Selection screen supporting T1M-2 module diagnostics.

To display a port Echo Canceller Selection screen for T1M-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M-2 module.

3. Select **Port**.

4. Select **Port 1** or **Port 2**, as applicable.

5. Select **Echo Canceller Channels**. The Echo Canceller Selection screen for the module port appears, as shown in **Figure 22**.

```
                                                                    [Bay 1]
       Echo Canceller Selection [Port 1]
       ===================================
       Channel #1       Channel #16
       Channel #2       Channel #17
       Channel #3       Channel #18
       Channel #4       Channel #19
       Channel #5       Channel #20
       Channel #6       Channel #21
       Channel #7       Channel #22
       Channel #8       Channel #23
       Channel #9       Channel #24
       Channel #10      Channel #25
       Channel #11      Channel #26
       Channel #12      Channel #27
       Channel #13      Channel #28
       Channel #14      Channel #29
       Channel #15      Channel #30

       <. Back
       X. Exit
       Enter selection: █
```

The **Channel #1** selection is highlighted upon entry to this screen.

The number of channels displayed varies according to the port type, as follows:

- T1 = 24 channels (12 on each column)

- T1/PR I = 23 channels (12 on left column, 11 on right column)

- E1/PRI = 30 channels (15 on each column)

After using the ARROW keys to select the channel you want, press **Enter** to go to the selected Echo Canceller Channel screen.

PORT 1 OR 2 – ECHO CANCELLER SELECTION – ECHO CANCELLER STATUS

The following section describes the port Echo Canceller Status screen supporting T1M-2 module diagnostics. The Echo Canceller Status screen displays the status information for the selected echo canceller. This screen refreshes periodically.

To display a port Echo Canceller Status screen for T1M-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M-2 module.

3. Select **Port 1** or **Port 2**, as applicable.

4. Select **Echo Canceller Channels**.

5. Select the channel to be diagnosed. The Echo Canceller Status screen appears for the channel selected, as shown in **Figure 23**.

```
[Bay 1] [Port 1] [Channel # 1]                    Arrow keys - Change channel
                                                   '.' - To exit screen
Incoming call

                                      | Span Side          | System Side
Echo Canceller Profile:               | Medium Echo        | Medium Echo
Companding Type:                      | uLaw               | uLaw
Adaptive Filter Operation:            | Active             | Active
Echo Canceller Converged:             | True               | True
Number of Echo Path Changes:          | 0                  | 0
Tone Disabler Mode:                   | On                 | On
Tone Disabler Status:                 | Enabled            | Enabled
Current Echo Return Loss:             | 0  dB              | 0  dB
Maximum Echo Return Loss:             | 0  dB              | 0  dB
Current Echo Return Loss Enhancement: | 0  dB              | 0  dB
Maximum Echo Return Loss Enhancement: | 0  dB              | 0  dB
Current Echo Delay:                   | 0  ms              | 0  ms
Max Echo Delay:                       | 0  ms              | 0  ms
Rin Level:                            | 0  dBm0            | 0  dBm0
Sin Level:                            | 0  dBm0            | 0  dBm0
Rin Applied Gain:                     | 0  dB              | 0  dB
Sout Applied Gain:                    | 0  dB              | 0  dB
Comfort Noise Level:                  | 0  dB              | 0  dB
```

The Echo Canceller Status screen for T1M-2 module diagnostics shows the following elements:

• The current channel number is highlighted upon entry to this screen.

• All physical devices on the system are associated with a given echo profile.

In addition to the general navigation guidelines described in "Navigating Through the Digital Trunk Diagnostics Screens" on **page 18-1033**, pressing the "." key transfers control back to the Echo Canceller Selection Screen.

DSP AUDIO DIAGNOSTICS

The following section describes the DSP (Digital Signal Processor) Audio Diagnostics Capture screen supporting T1M-2 module diagnostics.

To display the DSP Audio Diagnostics screen for T1M-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M-2 module.

3. Select **DSP Audio Diagnostics**. The Diagnostics Capture screen appears, as shown in **Figure 24**.

```
Diagnostics Capture                                          [Bay 1]

Current Status: [Idle/Capturing/Saving Capture]

Progress : X / [16/120]
Type     : [16/120] Seconds [Lite] Capture, [Trunk/System] Side
Channel  : Port 1, Channel 1

Current Setup:

Type     : [16/120] Seconds [Lite] Capture, [Trunk/System] Side
Channel  : Port 1, Channel 1 (Current Channel Status: [Inactive/Active])

Last Capture Summary: [Success/[Failed, [User Aborted/Call Aborted]]]

Type     : [16/120] Seconds [Lite] Capture, [Trunk/System] Side
Channel  : Port 1, Channel 1
Time     : xx/xx/xxxx, xx:xx:xx
Filename : date_time_trunkside_port_1_channel_1_16s_lite.bin

S - Setup | C - Start Capture | A - Abort Active Capture | X - Exit
```

A DSP on the Dual T1/E1/PRI Module provides echo canceller functionality and provides a way to capture DSP diagnostics on a specified channel. The capture can then be used to diagnose, troubleshoot, and debug echo problems that may occur on trunks connected to a T1M-2 module.

To obtain a DSP audio diagnostics capture:

1. Identify the extension of the endpoint on which an active call has poor quality audio, such as echo. See **Figure 25**.

2. Identify the extension of the trunk on which the call exists. See **Figure 25**.

a. Using Diagnostics Monitor, generate a Manual Dump of Call Diagnostics. Connect a System Monitor to the MiV

oice Office 250 in question.

b. From System Monitor select Dump Resource Managers and then Call Diagnostics. Search the log for the extens

i
o
n

i
n

q
u
e
s
ti
o
n
.

```
Diagnostics Monitor - [System Monitor - 172.17.158.193]
File  System Monitor  Window  Help

New Connection    Save to File...  Log to File...    Clear System Monitor Buffer      Pause System Monitor Buffer

-01:259- > Manual Dump of Call Diagnostics
-01:259- > ----------------------------
-01:259- > Call 1:
-01:259- >    <DB-SRVR>    1:PP009              <-- CONN    -->    <DB-SRVR>    1:PP006
-01:259- > Call 2:
-01:259- >    <DIG-EP>     1:1001               <-- CONN    -->    <SL-EP>      1:10220
-01:259- > Call 3:
-01:259- >    <LS-TRK>     1:94000              <-- CONN    -->    <DIG-EP>     1:1003
*** Command Completed ***



For Help, press F1
```

3. Identify the Trunk Hardware Address. Using the same System Monitor session in Diagnostics Monitor, dump the Hardware System by selecting System Monitor – Dump Resource Managers – Hardware System. Locate the extension of the trunk. In front of the extension you can locate the hardware address in the format [Board . Port . Device]. See **Figure 26**.

```
Diagnostics Monitor - [System Monitor - 172.17.158.193]
File  System Monitor  Window  Help

New Connection    Save to File...  Log to File...    Clear System Monitor Buffer      Pause System Monitor Buffer

-01:266- >       hw_port_sdb (LOOP START, [2.2.0]), number of devices = 1
-01:266- >          hw_device_sdb (LOOP START, [2.2.1], ID(1:'94001')), APP device type (123)
-01:266- >    hw_board_sdb (PRI Board, [3.0.0]), number of ports = 1
-01:266- >       hw_port_sdb (T1 PRIMARY RATE, [3.1.0]), number of devices = 23
-01:266- >          hw_device_sdb (B-CHANNEL TRUNK, [3.1.1], ID(1:'94002')), APP device type (126)
-01:266- >          hw_device_sdb (B-CHANNEL TRUNK, [3.1.2], ID(1:'94003')), APP device type (126)
-01:266- >          hw_device_sdb (B-CHANNEL TRUNK, [3.1.3], ID(1:'94004')), APP device type (126)
-01:266- >          hw_device_sdb (B-CHANNEL TRUNK, [3.1.4], ID(1:'94005')), APP device type (126)
-01:266- >          hw_device_sdb (B-CHANNEL TRUNK, [3.1.5], ID(1:'94006')), APP device type (126)
-01:266- >          hw_device_sdb (B-CHANNEL TRUNK, [3.1.6], ID(1:'94007')), APP device type (126)
-01:266- >          hw_device_sdb (B-CHANNEL TRUNK, [3.1.7], ID(1:'94008')), APP device type (126)
-01:266- >          hw_device_sdb (B-CHANNEL TRUNK, [3.1.8], ID(1:'94009')), APP device type (126)
-01:266- >          hw_device_sdb (B-CHANNEL TRUNK, [3.1.9], ID(1:'94010')), APP device type (126)
-01:266- >          hw_device_sdb (B-CHANNEL TRUNK, [3.1.10], ID(1:'94011')), APP device type (126)
-01:266- >          hw_device_sdb (B-CHANNEL TRUNK, [3.1.11], ID(1:'94012')), APP device type (126)
-01:266- >          hw_device_sdb (B-CHANNEL TRUNK, [3.1.12], ID(1:'94013')), APP device type (126)
-01:266- >          hw_device_sdb (B-CHANNEL TRUNK, [3.1.13], ID(1:'94014')), APP device type (126)
-01:266- >          hw_device_sdb (B-CHANNEL TRUNK, [3.1.14], ID(1:'94015')), APP device type (126)

For Help, press F1
```

4. Access the DSP Diagnostics Capture through the DMU. Using the DMU, start the capture. Using an SSH client (such as PuTTY), log into the MiVoice Office 250 using the 'it5k'

username. Select the Digital Trunk Diagnostics as shown in
Figure 27 on **page 1057**.

5. Select the bay identified in step 2 on **page 18-1053**. In the example shown in **Figure 27**, this is bay 3. Only the T1M-2 (Dual T1/E1/PRI Module) can perform a DSP Diagnostics Capture. BRM-2, LSM-2, LSM-4, and T1M do not have this functionality.

```
        Digital Trunk Diagnostics
        =========================
        A. Bay 1 - N/A [empty]
        B. Bay 2 - N/A [LSM4]
        C. Bay 3 - T1M-2
        X. Exit
        Enter selection: █




        System Reference Clock: Internal (Motherboard)
```

6. Select the **DSP Audio Diagnostics** option from the T1M-2 main menu, as shown in **Figure 28**.

```
                                                          [Bay 1]

        Dual T1/E1/PRI Module
        =====================
        A. Status
        B. Busy Out
        C. Version
        D. Port 1 [T1]
        E. Port 2 [T1/PRI]
        F. DSP Audio Diagnostics
        G. Clock Frequency
        H. Timers
        <. Back
        X. Exit
        Enter selection: █
```

7. Set up the capture. Press 'S' on the Diagnostics Capture screen (see Figure 24 on **page 1055**) to setup the capture.

8. Start the capture. Upon returning to the Diagnostics Capture screen, make sure the current channel status is active in the Current Setup section. You cannot start a capture on an inactive channel (i.e., there is no call currently in progress on the channel). Then press 'C' to start the capture. The Current Status section should change to show "Capturing" and the progress counter should start. You may abort a capture while the current status is "Capturing" by pressing 'A'. However, once the current status changes to "Saving Capture",

you can no longer abort the capture. Once the capture has been saved, the Last Capture Summary is updated. Please note the filename of the capture.

9. Retrieve the capture. There are two ways to retrieve the capture -- either by downloading it from the MiVoice Office 250 Web page, or by doing an ADD freeze. Assuming you have ADD setup, do a freeze from DB Programming, Diagnostics Monitor, or from an administrator endpoint. To retrieve the DSP diagnostics capture from the Web page, log into the Web page and navigate to Logging – Log Files. Retrieve the file under the Dual T1/E1/PRI Module section with the filename you got in step 6. If you retrieved the capture through the web page, attach it to the Remedy ticket and Mitel technical support can use this to troubleshoot your echo problem further. An ADD freeze automatically attaches the capture to a Remedy ticket.

DSP AUDIO DIAGNOSTICS – SETUP

The following section describes the DSP (Digital Signal Processor) Audio Diagnostics Capture screen supporting T1M-2 module diagnostics.

To display the DSP Audio Diagnostics screen for T1M-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M-2 module.

3. Select **DSP Audio Diagnostics**.

4. Select **S - Setup** from the lower left corner of the screen. The Setup DSP Diagnostics Capture screen appears, as shown in **Figure 29**.

```
                                                                    [Bay 1]
Setup DSP Diagnostics Capture

Question         Choice 1
                 Choice 2, etc.

Selection: _

[Enter] with no selection uses current setting
[.] [Enter] to exit
_____
Current Settings:

Capture Length: 16 seconds
Capture Type  : Lite
Echo Canceller: Trunk Side
Port number   : 1
Channel number: 1
```

To set up DSP Audio Diagnostics:

1. In the **Selection** prompt, type 1 or 2 for each of the following five questions, and then type "." and press **Enter**. Pressing **Enter** without typing a selection confirms the current setting.

   The program updates and returns to the Diagnostics Capture screen after you answer each question.

   - Capture Length
     - 16 seconds (default)
     - 120 seconds
   - Capture Type
     - Lite (default)
     - Extended – Select only if instructed to do so by Mitel Technical Support.
   - Echo Canceller
     - Span Side (default) – Select if you believe the echo is coming in from the trunk; that is, the local user hears the echo.
     - System Side – Select if you believe the echo is coming from the local system; that is, the distant user hears the echo.
   - Port number
     - 1
     - 2
   - Channel number

     23/24/30 – The number of channels is determined by the parameters of the trunk connected to the selected port

2. If you enter an invalid input, the screen prompts you again to answer the same question and indicates that the last input was invalid. The program continues to ask for a valid setting until you enter a valid setting, type a ".", and press **Enter**.

CLOCK FREQUENCY

The following section describes the Clock Frequency screen supporting T1M-2 module diagnostics. This screen refreshes periodically.

To display the DSP Audio Diagnostics screen for T1M-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M-2 module.

3. Select **Clock Frequency**. The CLOCK FREQUENCY screen appears, as shown in **Figure 30**.

```
[Bay 1]                                            '.' - To exit screen

                    << C L O C K    F R E Q U E N C Y >>

                        | Port 1          | Port 2
        ----------------+----------------+---------------
        Span Type       | T1              | E1/PRI
        ----------------+----------------+---------------
        Red Alarm       | No              | No
        RX Loss of Signal | No            | No
        Sync            | Yes             | Yes
        Frame Errors    | 2               | 2
        Slips           | 0               | 0
        ----------------+----------------+---------------
        Clock Timing    | Slave           | Master
        Providing Clock Ref | No          | No
        ----------------+----------------+---------------
        Simulated RX Freq | 1.533939 MHz  | 2.048000 MHz
        ----------------+----------------+---------------
```

The simulated frequencies are calculated assuming that the software timing is 100% accurate and that the TX frequency is 2.048 MHz.

The **Slips** and **Frame Errors** counts are cleared upon entrance to this screen.

When a port is unequipped, the **Span Type** is the only field displayed for that port. All other fields are blank.

When a port's RX Loss of Signal field reads **Yes**, the **Yes** is in reverse video to capture the user's attention, and the Simulated RX Freq field is blank.

TIMERS – TIMERS SELECTION

The following section describes the Timers Selection screen supporting T1M-2 module diagnostics.

To display the Timers Selection screen for T1M-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M-2 module.

3. Select **Timers**. The Timers Selection screen appears, as shown in **Figure 31**.
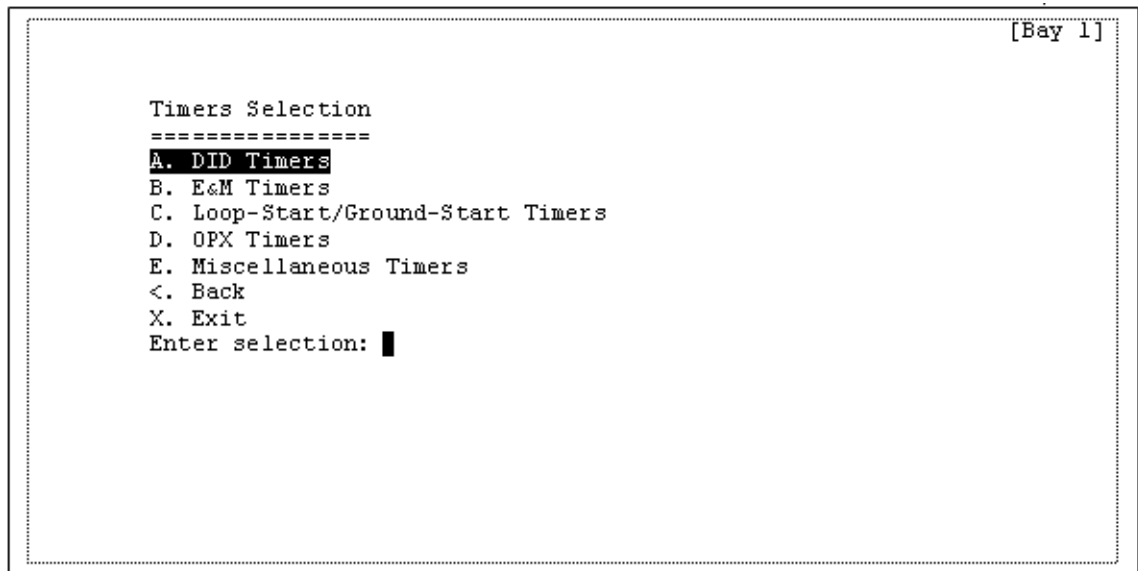
```
                                                              [Bay 1]


        Timers Selection
        ================
        A. DID Timers
        B. E&M Timers
        C. Loop-Start/Ground-Start Timers
        D. OPX Timers
        E. Miscellaneous Timers
        <. Back
        X. Exit
        Enter selection: █
```

DID Timers is highlighted upon entry to this screen.

Typing the selection shortcuts—A or a, B or b, <, X or x—activates the corresponding selection.

Pressing **Enter** activates the highlighted selection.

The cursor is located at the **Enter selection** prompt upon entry to this screen. After the UP ARROW or DOWN ARROW has been pressed, from that time forward, the cursor is located at the end of the highlighted selection.

TIMERS – TIMERS SELECTION – DID TIMERS

The following section describes the DID Timer Information screen that supports T1M-2 module diagnostics. This screen refreshes periodically.

To display the DID Timer Information screen for T1M-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M-2 module.

3. Select **Timers**.

4. Select **DID Timers**. The DID Timers Information screen appears, as shown in **Figure 32**.

```
                                                                   [Bay 1]
DID Timer Information:
Off-Hook Debounce:                              10 ms
On-Hook Debounce:                               10 ms
Disconnect Recognition:                         150 ms
Seizure Recognition:                            30 ms
Post Seize Delay:                               65 ms
Pre-Signal Delay:                               100 ms
Signal Hold:                                    200 ms
Post Signal Delay:                              30 ms
Pulse-Dial Inter-Digit Pause Recognition:       240 ms
Ready Timeout:                                  4000 ms




_____
Press '.' - To exit
Press any-other-key - To display the E&M Timers screen
```

These timers are programmable through DB programming and affect the behavior of T1-DID channels.

TIMERS – TIMERS SELECTION – E&M TIMERS

The following section describes the E&M Timer Information screen that supports T1M-2 module diagnostics. This screen refreshes periodically.

To display the E&M Timer Information screen for T1M-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M-2 module.

3. Select **Timers**.

4. Select **E&M Timers**. The E&M Timers Information screen appears, as shown in **Figure 33**.

```
                                                                    [Bay 1]
E&M Timer Information:
Transmit Handshake Delay:    100 ms        Receive Handshake Delay:   20 ms
Hookflash Duration:          600 ms        Hookflash Recognition:     300 ms
Ready Timeout:               4000 ms       Handshake Timeout:         5000 ms
Off-Hook Debounce:           10 ms         On-Hook Debounce:          10 ms
Dial Delay:                  70 ms         Dial Delay Hold:           140 ms
Wink Hold:                   215 ms        Wink Timeout:              350 ms
Seizure Debounce:            5 ms          Post Seize Delay:          65 ms
Post Signal Delay:           30 ms         Dial Tone Wait:            1000 ms
Short Pause Digit:           3000 ms       Answer Recognition:        4500 ms

False Signal Debounce:                     50 ms
Disconnect Recognition:                    1500 ms
Disconnect Flash Duration:                 15000 ms
Dialing Wait After Hookflash:              3000 ms
Disconnect Wait After Dialing:             3000 ms
Pulse-Dial Pulse Hold:                     60 ms
Pulse-Dial Inter-Pulse Pause:              40 ms
Pulse-Dial Inter-Digit Pause:              700 ms
Pulse-Dial Inter-Digit Pause Recognition:  300 ms
_____
Press '.' - To exit
Press any-other-key - To display the Loop-Start/Ground-Start Timers screen
```

These timers are programmable through the DB Programming interface and affect the behavior of T1-E&M channels.

TIMERS – TIMERS SELECTION – LOOP-START/GROUND-START TIMERS

The following section describes the Loop-Start/Ground-Start Timer Information screen that supports T1M-2 module diagnostics. This screen refreshes periodically.

To display the Loop-Start/Ground-Start Timer Information screen for T1M-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M-2 module.

3. Select **Timers**.

4. Select **Loop-Start/Ground-Start Timers**. The Loop-Start/Ground-Start Timers Information screen appears, as shown in **Figure 34**.

```
                                                              [Bay 1]
Loop-Start/Ground-Start Timer Information:
Pulse-Dial Pulse Hold:          60 ms
Pulse-Dial Inter-Pulse Pause:   40 ms
Tip-Ground Debounce:            50 ms
Ground-Start Transition:        10 ms
CO-to-CO Disconnect Debounce:   350 ms
IC-to-CO Disconnect Debounce:   600 ms
Dialing Disconnect Debounce:    2000 ms
Hookflash Duration:             600 ms
Minimum No-Ring:                6000 ms
Minimum Ring:                   500 ms
Dialing Wait After Hookflash:   3000 ms
Disconnect Wait After Dialing:  20000 ms
Loop-Start Dial Delay:          1500 ms
Pulse-Dial Inter-Digit Pause:   700 ms
Short Pause Digit:              3000 ms
Ground-Start Dial Delay:        3000 ms




_____
Press '.' - To exit
Press any-other-key - To display the OPX Timers screen
```

These timers are programmable through the DB Programming interface and affect the behavior of T1 Loop-Start/Ground-Start channels.

TIMERS – TIMERS SELECTION – OPX TIMERS

The following section describes the OPX (Off-Premises Extension) Timer Information screen that supports T1M-2 module diagnostics. This screen refreshes periodically.

To display the OPX Timer Information screen for T1M-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M-2 module.

3. Select **Timers**.

4. Select **OPX Timers**. The **OPX** Timer Information screen appears, as shown in **Figure 35**.

```
                                                                    [Bay 1]
OPX Timer Information:
Off-Hook Debounce:                                   10 ms
On-Hook Debounce:                                    10 ms
Disconnect Recognition:                              700 ms
Seizure Recognition:                                 150 ms
Hookflash Recognition:                               300 ms
Post Seize Delay:                                    65 ms
Pulse-Dial Inter-Digit Pause Recognition:            300 ms
Disconnect Flash Duration:                           1000 ms
Ground Key Debounce:                                 30 ms
```

```
Press '.' - To exit
Press any-other-key - To display the Board Timers screen
```

These timers are programmable through the DB Programming interface and affect the behavior of T1 OPX channels.

TIMERS – TIMERS SELECTION – MISCELLANEOUS TIMERS

The following section describes the Miscellaneous Timer Information screen that supports T1M-2 module diagnostics. This screen refreshes periodically.

To display the Miscellaneous Timer Information screen for T1M-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M-2module.

3. Select **Timers**.

4. Select **Miscellaneous Timers**. The Miscellaneous Timer Information screen appears, as shown in **Figure 36**.

```
                                                                [Bay 1]
Miscellaneous Timer Information:
Loopback Timeout:   600 secs

















Press '.' - To exit
Press any-other-key - To display the DID Timers screen
```

These timers are programmable through the DB Programming interface.

T1/E1/PRI MODULE (T1M) DIAGNOSTICS SCREENS

The following sections describe the DMU screens that support the single-port T1E1/PRI Module diagnostics. This screen periodically refreshes option D.

MAIN

This section describes the main screen for single-port T1/E1/PRI Module (T1M) diagnostics.

To display the T1M main screen:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. From the Module Selection Screen select a **Bay** containing a T1M module. The main screen for the T1M module appears, as shown in **Figure 37**.

```
                                                              [Bay 1]

         T1/E1/PRI Module
         ================
         A. Status
         B. Busy Out
         C. Version
         D. Port [E1/PRI]
         E. Timers
         <. Back
         X. Exit
         Enter selection: █
```

The **Status** selection is highlighted upon entry to this screen.

If the port is [UNEQUIPPED], the Status, Busy Out, and Timers selections are not valid.

The port type, enclosed in brackets, values include:

- T1

- T1/PRI

- E1/PRI

- UNEQUIPPED

STATUS

The following section describes the STATUS screen for the T1M module. This screen refreshes periodically.

To display the Status screen for T1M diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. From the Module Selection Screen select a **Bay** containing a T1M module.

3. From the T1M main screen, select **A. Status**. The STATUS screen appears, as shown in **Figure 38**.

```
[Bay 1]                  << S T A T U S >>            's' - SYNC Status screen
                                                     '.' - To exit screen

   Port Type        SYNC Status          REF LED        MSTR LED
   T1               INIT-TRBL/RED-ALARM   OFF            OFF


   Channel     |                 1 1 1 1 1 1 1 1 1 1 2 2 2 2
   Number      | 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4
   Status      | + + + + + + + + + + + + + + + + + + + + + + 0 0


   Legends: B - Busied out                  + - Idle
            i - Busying out incoming call    X - Offline
            o - Busying out outgoing call    - - Unequipped
            I - Incoming call
            O - Outgoing call
```

The Status screen for T1M module diagnostics shows the elements described in the following paragraphs.

The **Port Type** field appears one of the following states:

- T1
- T1/PRI

- E1/PRI
- UNEQUIPPED

The **SYNC Status** field displays one of the following values:

- RED-ALARM
- INIT-TRBL
- INIT-TRBL/RED-ALARM

- SYNC
- Blank, if the port is unequipped

| NOTE | The RED-ALARM, INIT-TRBL, INIT-TRBL/RED-ALARM values blinks to gain the user's attention. |

The **REF LED** and **BUSY LED** fields display a value of OFF, ON, or blank, if the port is unequipped.

Channel Number rows count to the maximum number of channels for the corresponding port type:

- 23 for T1/PRI
- 24 for T1
- 30 for E1/PRI

In addition to the general navigation guidelines described in "Navigating Through the Digital Trunk Diagnostics Screens" on **page 18-1033**, the following control applies:

Pressing the S key or the s key transfers control to the SYNC Status screen.

SYNC STATUS

The following section describes the SYNC STATUS screen for T1M (or T1M-2) modules. The SYNC STATUS screen shows the same information that the M8000 and M8001 System Message Print messages show but in a real-time format instead of a log format. For descriptions of the various Initial-Trouble/Red-Alarm conditions, refer to the *Message Print Diagnostics Manual*, part no. 550.8018. This screen refreshes periodically.

To display the Sync Status screen for T1M diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. From the Module Selection Screen select a **Bay** containing a T1M module.

3. From the T1M main screen, select **Status**.

4. While in the Status screen, press the S or s key. The SYNC STATUS screen appears, as shown in **Figure 39**.

```
[Bay 1]                 << S Y N C   S T A T U S >>         's' - Status screen
                                                            '.' - To exit screen

Port 1 (T1):                                        | Initial | Red   |
  Condition                                         | Trouble | Alarm |
    Receiver Loss of Signal from Network.......... |    *    |   *   |
    Transmitter Loss of Signal.................... |    *    |       |
    Receiving Yellow Alarm........................ |         |   *   |
    Receiving Blue Alarm.......................... |         |       |
    T1 Signal Out-of-Frame........................ |         |       |
    Severely Errored Seconds...................... |    *    |       |
    Transmitting Blue Alarm....................... |         |   *   |
    Remote Loopback State......................... |         |       |

Port 2 (E1/PRI):                                    | Initial | Red   |
  Condition                                         | Trouble | Alarm |
    Receiver Loss of Signal from Network.......... |    *    |   *   |
    Transmitter Loss of Signal.................... |    *    |       |
    Loss of Frame Synchronization................. |         |       |
    Loss of Multiframe Synchronization............ |         |       |
    Alarm Indication Signal (AIS) Reception....... |         |       |
    Remote Alarm Indication Signal (RAI) Reception |         |       |
    Transmitting AIS.............................. |         |       |
```

The Sync Status screen for T1M module diagnostics shows the following elements:

• The heading for each port displays the port type in parenthesis. Valid port types include:

  • T1

  • T1/PRI

  • E1/PRI

  • UNEQUIPPED

- The single-port T1M does not indicate a Port number.

- If a port is unequipped, that port won't show a corresponding SYNC status table.

- The tables for T1 and T1/PRI are different from the table for E1/PRI.

In addition to the general navigation guidelines described in "Navigating Through the Digital Trunk Diagnostics Screens" on **page 18-1033**, the following control applies:

Pressing the S key or the s key transfers control to the Status screen.

BUSY OUT

The following section describes the BUSY-OUT screen for the T1M module diagnostics. This screen refreshes periodically.

To display the BUSY-OUT screen for T1M diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. From the Module Selection Screen select a **Bay** containing a T1M module.

3. From the T1M main screen, select **Busy Out**. The BUSY-OUT screen for T1M appears, as shown in **Figure 40**.

```
[Bay 1]                    << B U S Y - O U T >>      't'/'u' - Tag/Untag
                                                       Arrow keys - Change channel
                                                       '.' - To exit screen


Port
  Channel    |                    1 1 1 1 1 1 1 1 1 1 2 2 2 2 2 2 2 2 2 2 3
  Number     | 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0
  Status     | + + B B + I + + + + + O + + + + + + + + + I I + + + + + O
  Busy Tag   |     *  *    *                 *

  Legends: B - Busied out                    + - Idle
           i - Busying out incoming call     X - Offline
           o - Busying out outgoing call     - - Unequipped
           I - Incoming call
           O - Outgoing call                 * - Tagged for busy out
```

The Busy Out screen for T1M module diagnostics shows the following elements:

- **Port 1** is highlighted and flashing upon entry to this screen.

- The **Channel Number** identifies the channels available with the trunk programmed for the module port.

- The **Instructions** summarize how to use the screen commands to busy-out channels or ports.

- The **Status** line identifies the status of the channel above it.

- The **Busy-out Tag** line identifies the busy-out condition of the channel above it.

- The **Legends** area defines the Status line and Busy-out Tag line symbols.

Instead of the navigation guidelines described in "Navigating Through the Digital Trunk Diagnostics Screens" on **page 18-1033**, the following controls apply:

- From the port selection, pressing the LEFT ARROW moves the cursor to the last channel in the port.

- From the port selection, pressing the RIGHT ARROW moves the cursor to the first channel in the port.

- Pressing the LEFT ARROW or RIGHT ARROW moves the cursor between channels in the port.

- Pressing the LEFT ARROW key from the first channel moves the cursor to the last channel in the port.

- Pressing the RIGHT ARROW from the last channel moves the cursor to the first channel in the port.

- Pressing the UP ARROW or DOWN ARROW while a channel is selected selects the port.

- If a port is selected, pressing the T key or the t key tags all channels in the port for busy-out.

- If a port is selected, pressing the U key or the u key untags all channels in the port for busy-out.

- If a channel is selected, pressing the T key or the t key tags the channel for busy-out.

- If a channel is selected, pressing 'the U key or the u key untags the channel for busy-out.

VERSION INFORMATION

The following section describes the Version Information screen for T1M module diagnostics.

To display the Version Information screen for T1M diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. From the Module Selection Screen select a **Bay** containing a T1M module.

3. From the T1M main screen, select **Version**. The Version Information screen appears, as shown in **Figure 41**.

```
Version Information                                            [Bay 1]
==================


T1 Application
  Version:          2.3
  Build Date:       Wed May 23 2007 11:23:58
  Build directory:  /home/ymc/workspace/RemoteOffice/intl/apps/t1



DSP
  Version:          0.0



Press any key to exit
```

The format of the version numbers varies in accordance with data that is available for each software/hardware module.

PORT

The following section describes the Port screen for T1M module diagnostics.

To display the Port screen for T1M diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. From the Module Selection Screen select a **Bay** containing a T1M module.

3. From the T1M main screen, select **Port**. The Port screen appears, as shown in **Figure 42**.

```
                                                                    [Bay 1]


        Port 1 [T1/PRI]
        ===============
        A. Layer 1
        B. Channels
        C. T1 Performance
        D. Call Statistics
        E. ISDN Diagnostics
        F. Loopback
        G. Echo Canceller Channels
        <. Back
        X. Exit
        Enter selection: █
```

The **Layer 1** selection is highlighted upon entry to this screen.

The screen title appears the corresponding port number and the port type in brackets. Port types include:

- T1

- T1/PRI

- E1/PRI

ISDN Diagnostics appears only when the port is programmed as a T1/PRI or E1/PRI trunk.

The **Performance** option changes to conform with the port type. That is, when the port is programmed as T1 or T1/PRI, the option appears as **T1 Performance**, and when the port is programmed as E1/PRI, the option appears as **E1 Performance**.

PORT – LAYER 1

The following section describes the Open Systems Interconnect (OSI) model Layer 1 screen for T1M module diagnostics. The LAYER-1 screen is useful to get a glance at a T1/E1/PRI port Layer-1 status and the channel activity on that port, all in one screen. This screen refreshes periodically.

To display the Layer 1 screen for T1M diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M module.

3. Select **Port**.

4. Select **Layer 1**. The LAYER-1 screen for the port appears, as shown in **Figure 43**.

```
[Bay 1] [Port 1]          << L A Y E R - 1 >>         'A' - Toggle Alarm Logging
                                                      '.' - To exit screen

Span Type............ El-PRI                           Red Alarm..... No
Card Slot............ 1          TCR Local Loopback.. No   Frame Errors.. 0
Com. Chan. Signalling Yes        TCR Remote Loopback. No   Slips......... 0
CRC Enabled.......... Yes        FMR Remote Loopback. No   Sync.......... Yes
HDB3 Enabled......... Yes        FMR Local Loopback.. No   MF Sync....... Yes
Term. Resistor Jumper 120 ohms   TX Loss of Signal... No   CRC Sync...... Yes
TX Pulse Template.... 120 ohms   TX RAI.............. Auto  Si2........... 1
Clock Timing......... Slave      TX AIS.............. No    Sil........... 1
Ref. Clock Source.... ANCC       RX Loss of Signal... No    FEBEs......... 0
System Sync Source... This El    RX RAI.............. No     CRC Errors.... 0
Local Sync Request... Yes        RX AIS.............. No
Clock Mode........... CEPT       Network Loopback.... N/A   Alarm Logging. Yes

B-CHANNEL|              1 1 1 1 1 1 1 1 1 1 2 2 2 2 2 2 2 2 2 2 3
NUMBER   | 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0
STATUS   |
```

This screen displays a port-number field at the top of the screen.

PORT – CHANNEL SELECTION

The following section describes the port Channel Selection screen on T1M module diagnostics.

To display the Channel Selection screen for T1M diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. From the Module Selection Screen select a **Bay** containing a T1M module.

3. From the T1M main screen, select **Port**.

4. From the Port screen, select **Channels**. The Channel Selection screen for the port appears, as shown in **Figure 44**.

```
                                                              [Bay 1]
        Channel Selection [Port 1]
        ==========================
        Channel #1         Channel #16
        Channel #2         Channel #17
        Channel #3         Channel #18
        Channel #4         Channel #19
        Channel #5         Channel #20
        Channel #6         Channel #21
        Channel #7         Channel #22
        Channel #8         Channel #23
        Channel #9         Channel #24
        Channel #10        Channel #25
        Channel #11        Channel #26
        Channel #12        Channel #27
        Channel #13        Channel #28
        Channel #14        Channel #29
        Channel #15        Channel #30

        <. Back
        X. Exit
        Enter selection: █
```

The **Channel #1** selection is highlighted upon entry to this screen.

The number of channels displayed varies according to the port type, as follows:

- T1 = 24 channels (12 on each column)

- T1/PR I = 23 channels (12 on left column, 11 on right column)

- E1/PRI = 30 channels (15 on each column)

After using the ARROW keys to select the channel you want, press **Enter** to go to the selected Channel screen.

PORT – CHANNEL SELECTION – CHANNEL

The following section describes the port Channel screen that supports T1M module diagnostics. This screen refreshes periodically.

To display the port Channel screen for T1M diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M module.

3. Select **Port**.

4. Select **Channels**.

5. Select the channel you want to diagnose. The Channel screen for the selected channel appears, as shown in **Figure 45**.

```
[Bay 1] [Port 1] Channel #1                 Arrow keys - Move
                                            'c' - Toggle Diag. Logging
CO Line Type:          Unknown              '.' - To exit screen
Current State:         Non-Functional
Current Substate:      Non-Functional Entry
Next Substate:         Global Relay Delay
Response Buffer:       $62 $1A $01 $3C $00 $0A
Static Line Type:      %00000000
Dynamic Line Type:     %00000000
Last Digit Dialed:     $00
TX Gain:               %00000000
RX Gain:               %00000000
Clear Channel:         No
Miscellaneous Flags:   %00000000
Rockwell Address:      $02
T1 Channel:            $01
RX Signalling:         A=1 B=1
TX Signalling:         A=0 B=0
Channel LED:                        CO Task Diagnostics Logging:  Yes
```

For non-ISDN channels, a **CO Task Diagnostics Logging** field appears in the lower right corner. This field can be switched between Yes and No by pressing the C key or the c key. The output controlled by this field goes into the T1/DT1/BRI Diagnostics log.

PORT – T1 PERFORMANCE

The following section describes the port T1 PERFORMANCE screen supporting T1M module diagnostics. This screen is useful to view the same Hourly and Daily error counts that are available through the DB Programming. This screen refreshes periodically.

To display a port T1 PERFORMANCE screen for T1M diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M module.

3. Select **Port**.

4. Select **T1 Performance**. The T1 PERFORMANCE screen for the port appears, as shown in **Figure 46**.

```
11:23:58 Wednesday, May 23, 2007                                    [Bay 1]

                    << T 1   P E R F O R M A N C E >>

Port #1
                         Current Counts          Error Thresholds
T1 Performance Parameter    Hourly     Daily       Hourly     Daily
Bipolar Violations              0       1436        53360    133400
CRC Errors                      0         13        53184    132960
Out-of-Frame Errors             0         91            7        17
Frame Slip Errors              91      61733            2         4
Errored Seconds                 0          4          259       648
Severely Errored Seconds        0          0           40       100
Unavailable Seconds             0          0           10        10




Press any key to exit
```

This screen shows only the counts for the current hour and current day, as opposed to the DB Programming data, which shows a history of 24 hourly counts and a history of seven daily counts for each T1 Performance parameter.

PORT – CALL STATISTICS

The following section describes the port CALL STATISTICS screen supporting T1M module diagnostics. This screen refreshes periodically.

To display a port CALL STATISTICS screen for T1M diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M module.

3. Select **Port**.

4. Select **Call Statistics**. The CALL STATISTICS screen for the port appears, as shown in **Figure 47**.

```
[Bay 1] [Port 1]    << C A L L   S T A T I S T I C S >>    'c' - To clear counts
                                                           '.' - To exit screen
Current Time:      11:23:58 Wednesday, May 23, 2007
Last Time Cleared: 11:23:58 Wednesday, May 23, 2007


            Incoming   Outgoing                       Incoming   Outgoing
Channel       Calls      Calls           Channel        Calls      Calls
   1            0          0                 13            0          0
   2            0          0                 14            0          0
   3            0          0                 15            0          0
   4            0          0                 16            0          0
   5            0          0                 17            0          0
   6            0          0                 18            0          0
   7            0          0                 19            0          0
   8            0          0                 20            0          0
   9            0          0                 21            0          0
  10            0          0                 22            0          0
  11            0          0                 23            0          0
  12            0          0

    Total # of Connected Calls:  0
```

The call counts are reset to zero when the module powers up; therefore the counts are not preserved across module/system reset.

This screen displays 23 channels for T1/PRI, 24 channels for T1, or 30 channels for E1/PRI ports.

PORT – ISDN DIAGNOSTICS

The following section describes the port ISDN Diagnostics screen supporting T1M module diagnostics.

To display a port ISDN Diagnostics screen for T1M diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M module.

3. Select **Port**.

4. Select **ISDN Diagnostics**. The ISDN Diagnostics screen for the selected port appears, as shown in **Figure 48**.

```
                                                           [Bay 1]


       ISDN Diagnostics [Port 1]
       =========================
       A. Data Link Status
       B. D-Channel Diagnostics
       <. Back
       X. Exit
       Enter selection: █
```

The **Data Link Status** selection is highlighted upon entry to this screen.

Typing the selection shortcuts—A or a, B or b, <, X or x—activates the corresponding selection.

Pressing **Enter** activates the highlighted selection.

The cursor is located at the **Enter selection** prompt upon entry to this screen. Once the UP ARROW or DOWN ARROW has been pressed, from that time forward, the cursor is located at the end of the highlighted selection.

PORT – ISDN DIAGNOSTICS – DATA LINK STATUS

The following section describes the port ISDN Data Link Status screen supporting T1M module diagnostics. This screen is useful to troubleshoot OSI model Layer 2 problems on an ISDN connection. The screen refreshes periodically.

To display a port ISDN Data Link Status screen for T1M diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M module.

3. Select **Port**.

4. Select **ISDN Diagnostics**.

5. Select **Data Link Status**. The ISDN Data Link Status screen for the module port appears, as shown in **Figure 49**.

```
[Bay 1] Port #1                                        '.'  - To exit screen

PRI (ISDN) Configuration
Switch Type... DMS-100          K.......... 7
T200......... 1000 ms           N200....... 3
T203......... 10000 ms          N201....... 260
T204......... 2000 ms           N204....... 3

Layer-2 Network Mode:  DISABLED

Data Link Control Block
D-Channel Status... UP
Sapi.............. $00
Tei............... $00
State............. MULTIPLE FRAME ESTABLISHED
VA................ $37
VS................ $37
VR................ $3B

L2_Queue_Usage.... $00
L2_Queue_Usage_Max $00
```

PORT – ISDN DIAGNOSTICS – D-CHANNEL DIAGNOSTICS

The following section describes the port D-Channel Diagnostics screen supporting T1M module diagnostics. The D-Channel Diagnostics screen allows you to enable or disable the D-Channel Diagnostics for the current port. This screen refreshes periodically.

To display the D-Channel Diagnostics screen for T1M diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M module.

3. Select **Port**.

4. Select **ISDN Diagnostics**.

5. Select **D-Channel Diagnostics**. The ISDN D-Channel Diagnostics screen for the module port appears, as shown in **Figure 50**.

```
                                                              [Bay 1]


        D-Channel Diagnostics: Port 1: [DISABLED]
        ========================================
        A.  Enable D-Channel Diagnostics (Level-1)
        B.  Enable D-Channel Diagnostics (Level-2)
        C.  Disable D-Channel Diagnostics
        D.  Display D-Channel Diagnostics
        <.  Back
        X.  Exit
        Enter selection:
```

The **Enable D-Channel Diagnostics (Level-1)** selection is highlighted upon entry to this screen.

For the T1M no reference to Port # appears in the menu title. The state of the current port's D-Channel Diagnostics is enclosed in brackets and can have one the following values:

- DISABLED

- LEVEL-1

- LEVEL-2

- UNDEFINED

The **Display D-Channel Diagnostics** option has no effect if the D-Channel Diagnostics is disabled.

PORT – ISDN DIAGNOSTICS – D-CHANNEL DIAGNOSTICS – DISPLAY

The following section describes the port D-Channel Diagnostics Display screen supporting T1M module diagnostics. The D-Channel Diagnostics Display screen allows you to view the D-Channel Diagnostics in real-time.

To display a port D-Channel Diagnostics Display screen for T1M diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M module.

3. Select **Port**.

4. Select **D-Channel Diagnostics**.

5. Select **Enable D-Channel Diagnostics Level-1** or **Level-2**, as applicable.

6. Select **Display D-Channel Diagnostics**. The D-Channel Diagnostics Display screen for the module port appears, as shown in **Figure 51**.

```
                                                                      [Bay 1]

D-Channel Diagnostics: Port 1: [LEVEL-1]
========================================

Pressing any key now will start the D-channel Diagnostics real-time display on
the screen.  Once the D-Channel Diagnostics real-time display has started,
pressing the '.' key will exit back to D-Channel Diagnostics menu.

Press any key to continue: █
```

The state of the D-Channel diagnostics for the current port is displayed in brackets. Possible values include:

- LEVEL-1

- LEVEL-2

- UNDEFINED

PORT – LOOPBACK

The following section describes the port Loopback screen supporting T1M module diagnostics. The LOOPBACK screen allows you to put the current port into one of four loopback modes and to transmit and detect specific bit patterns. This screen refreshes periodically.

To display a port Loopback screen for T1M diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M module.

3. Select **Port**.

4. Select **Port 1** or **Port 2**, as applicable.

5. Select **Loopback**. The LOOPBACK screen for the module port appears, as shown in **Figure 52**.

```
[Bay 1] [Port #1]         << L O O P B A C K >>      SPACE - Activate Selection
                                                     Arrow keys - Change Selection
                                                     '.' - To exit screen


Loopback Type        | Loopback Status
Digital Local        | DISABLED
Digital Remote       | DISABLED
Transceiver Local    | DISABLED
Transceiver Remote   | ENABLED (remotely)



Bit Pattern          | Pattern Transmit | Pattern Received
Loop-up              | DISABLED         | NO
Loop-down            | DISABLED         | NO
Framed 0101          | DISABLED         | ---
```

The Loopback screen for T1M module diagnostics shows the following elements:

- The **Digital Local** loopback is selected upon entry to this screen.

- The **Loopback Status** and **Pattern Transmit** fields display either DISABLED or flashing ENABLED. Additionally, the Transceiver Remote loopback type may also display a flashing ENABLED (remotely) value for its Loopback Status.

- The **Pattern Received** fields display either YES or NO, except for the Framed 0101 bit pattern which always appears as **---**.

- The **Bit Errors** field applies to the Framed 2^XX-1 bit patterns only and displays a number only when the corresponding bit pattern is active. The other bit patterns always display as **---**.

Instead of the navigation guidelines described in "Navigating Through the Digital Trunk Diagnostics Screens" on , the following controls apply:

- Pressing the UP ARROW or the LEFT ARROW moves the highlight to the next Loopback Type field or Bit Pattern field in the upward direction.

- Pressing the DOWN ARROW or the RIGHT ARROW moves the highlight to the next Loopback Type field or Bit Pattern field in the downward direction.

- Pressing the SPACEBAR activates the highlighted Loopback Type or Bit Pattern. When a Loopback Type or Bit Pattern is enabled, all the other Loopback and Bit Patterns are disabled and the corresponding Loopback Status or Pattern Transmit entry is enabled.

TIMERS – TIMERS SELECTION

The following section describes the Timers Selection screen that supports T1M module diagnostics. This screen refreshes periodically.

To display the Timers Selection screen for T1M diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M module.

3. Select **Timers**. The Timers Selection screen appears, as shown in **Figure 53**.

```
                                                              [Bay 1]


        Timers Selection
        ===============
        A. DID Timers
        B. E&M Timers
        C. Loop-Start/Ground-Start Timers
        D. OPX Timers
        E. Miscellaneous Timers
        <. Back
        X. Exit
        Enter selection: █
```

DID Timers is highlighted upon entry to this screen.

Typing the selection shortcuts—A or a, B or b, <, X or x—activates the corresponding selection.

Pressing **Enter** activates the highlighted selection.

The cursor is located at the **Enter selection** prompt upon entry to this screen. Once the UP ARROW or DOWN ARROW has been pressed, from that time forward, the cursor is located at the end of the highlighted selection.

TIMERS – TIMERS SELECTION – DID TIMERS

The following section describes the DID Timers Information screen that supports T1M module diagnostics. This screen refreshes periodically.

To display the DID Timers Information screen for T1M diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M module.

3. Select **Timers**.

4. Select **DID Timers**. The DID Timer Information screen appears, as shown in **Figure 54**.

```
                                                                    [Bay 1]
DID Timer Information:
Off-Hook Debounce:                              10 ms
On-Hook Debounce:                               10 ms
Disconnect Recognition:                         150 ms
Seizure Recognition:                            30 ms
Post Seize Delay:                               65 ms
Pre-Signal Delay:                               100 ms
Signal Hold:                                    200 ms
Post Signal Delay:                              30 ms
Pulse-Dial Inter-Digit Pause Recognition:       240 ms
Ready Timeout:                                  4000 ms




Press '.' - To exit
Press any-other-key - To display the E&M Timers screen
```

These timers are programmable through DB Programming and affect the behavior of T1-DID channels.

TIMERS – TIMERS SELECTION – E&M TIMERS

The following section describes the E&M Timer Information screen that supports T1M module diagnostics. This screen refreshes periodically.

To display the E&M Timer Information screen for T1M diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M module.

3. Select **Timers**.

4. Select **E&M Timers**. The E&M Timer Information screen appears, as shown in **Figure 55**.

```
                                                               [Bay 1]
E&M Timer Information:
Transmit Handshake Delay:   100 ms      Receive Handshake Delay:   20 ms
Hookflash Duration:         600 ms      Hookflash Recognition:     300 ms
Ready Timeout:              4000 ms     Handshake Timeout:         5000 ms
Off-Hook Debounce:          10 ms       On-Hook Debounce:          10 ms
Dial Delay:                 70 ms       Dial Delay Hold:           140 ms
Wink Hold:                  215 ms      Wink Timeout:              350 ms
Seizure Debounce:           5 ms        Post Seize Delay:          65 ms
Post Signal Delay:          30 ms       Dial Tone Wait:            1000 ms
Short Pause Digit:          3000 ms     Answer Recognition:        4500 ms

False Signal Debounce:                  50 ms
Disconnect Recognition:                 1500 ms
Disconnect Flash Duration:              15000 ms
Dialing Wait After Hookflash:           3000 ms
Disconnect Wait After Dialing:          3000 ms
Pulse-Dial Pulse Hold:                  60 ms
Pulse-Dial Inter-Pulse Pause:           40 ms
Pulse-Dial Inter-Digit Pause:           700 ms
Pulse-Dial Inter-Digit Pause Recognition:  300 ms
_____
Press '.' - To exit
Press any-other-key - To display the Loop-Start/Ground-Start Timers screen
```

These timers are programmable through DB Programming and affect the behavior of T1-E&M channels.

TIMERS – TIMERS SELECTION – LOOP-START/GROUND-START TIMERS

The following section describes the Loop-Start/Ground-Start Timer Information screen that supports T1M module diagnostics. This screen refreshes periodically.

To display the Loop-Start/Ground-Start Timer Information screen for T1M diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M module.

3. Select **Timers**.

4. Select **Loop-Start/Ground-Start Timers**. The Loop-Start/Ground-Start Timers Information screen appears, as shown in **Figure 56**.

```
                                                                    [Bay 1]
Loop-Start/Ground-Start Timer Information:
Pulse-Dial Pulse Hold:         60 ms
Pulse-Dial Inter-Pulse Pause:  40 ms
Tip-Ground Debounce:           50 ms
Ground-Start Transition:       10 ms
CO-to-CO Disconnect Debounce:  350 ms
IC-to-CO Disconnect Debounce:  600 ms
Dialing Disconnect Debounce:   2000 ms
Hookflash Duration:            600 ms
Minimum No-Ring:               6000 ms
Minimum Ring:                  500 ms
Dialing Wait After Hookflash:  3000 ms
Disconnect Wait After Dialing: 20000 ms
Loop-Start Dial Delay:         1500 ms
Pulse-Dial Inter-Digit Pause:  700 ms
Short Pause Digit:             3000 ms
Ground-Start Dial Delay:       3000 ms




_____
Press '.' - To exit
Press any-other-key - To display the OPX Timers screen
```

These timers are programmable through DB Programming and affect the behavior of T1 Loop-Start/Ground-Start channels.

TIMERS – TIMERS SELECTION – OPX TIMERS

The following section describes the OPX (Off-Premises Extension) Timer Information screen that supports T1M module diagnostics. This screen refreshes periodically.

To display the OPX Timer Information screen for T1M diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M module.

3. Select **Timers**.

4. Select **OPX Timers**. The **OPX** Timer Information screen appears, as shown in **Figure 57**.

```
                                                                    [Bay 1]
OPX Timer Information:
Off-Hook Debounce:                          10 ms
On-Hook Debounce:                           10 ms
Disconnect Recognition:                     700 ms
Seizure Recognition:                        150 ms
Hookflash Recognition:                      300 ms
Post Seize Delay:                           65 ms
Pulse-Dial Inter-Digit Pause Recognition:   300 ms
Disconnect Flash Duration:                  1000 ms
Ground Key Debounce:                        30 ms









Press '.' - To exit
Press any-other-key - To display the Board Timers screen
```

These timers are programmable through DB Programming and affect the behavior of T1-OPX channels.

TIMERS – TIMERS SELECTION – MISCELLANEOUS TIMERS

The following section describes the Miscellaneous Timer Information screen that supports T1M module diagnostics. This screen refreshes periodically.

To display the Miscellaneous Timer Information screen for T1M diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a T1M module.

3. Select **Timers**.

4. Select **Miscellaneous Timers**. The Miscellaneous Timer Information screen appears, as shown in **Figure 58**.

```
                                                                    [Bay 1]
Miscellaneous Timer Information:
Loopback Timeout:   600 secs

















Press '.' - To exit
Press any-other-key - To display the DID Timers screen
```

These timers are programmable through DB Programming.

BASIC RATE MODULE (BRM-2) DIAGNOSTICS SCREENS

The following sections describe the DMU screens that support the Basic Rate Module (BRM-2) diagnostics.

MAIN

This section describes the main screen for Basic Rate Module (BRM-2) diagnostics. This screen periodically refreshes options D and E.

To display the BRM-2 main screen:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a BRM-2 module. The main screen appears, as shown in **Figure 59**.

```
                                                                    [Bay 1]


        BRM-S Module
        ============
        A.  Status
        B.  Busy Out
        C.  Version
        D.  ISDN Diagnostics - Port 1 [Trunk]
        E.  ISDN Diagnostics - Port 2 [Trunk]
        <.  Back
        X.  Exit
        Enter selection:
```

The **Status** selection is highlighted upon entry to this screen.

If a port is [UNEQUIPPED], the corresponding ISDN Diagnostics selection is not valid.

The port type enclosed in brackets is either **TRUNK** or **UNEQUIPPED**.

STATUS

The following section describes the STATUS screen for the BRM-2 module. This screen refreshes periodically.

To display the Status screen for BRM-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a BRM-2 module.

3. Select **Status**. The STATUS screen appears, as shown in **Figure 60**.

```
[Bay 1]                     << S T A T U S >>              '.' - To exit screen


                    Port #1                Port #2

Configuration:      TE                     NOT EQUIPPED
Current State:      ACTIVATED

Status:
 Loopback Enabled:  NO
 Sync:              YES
 LOS:               NO

B1 Channel:         IDLE
 TX Gain(dB):       0
 RX Gain(dB):       0

B2 Channel:         IDLE
 TX Gain(dB):       0
 RX Gain(dB):       0
```

The "B<*n*> Channel" field is an improved version of the former "Bn Channel In Use" field, which displayed a "*" when the B-channel was busy on a call. Valid values for this field include:

- IDLE

- OUTGOING-CALL

- INCOMING-CALL

- BUSYING-OUT

- BUSIED-OUT

If a port is **NOT EQUIPPED**, the remainder of its corresponding fields are blank.

BUSY OUT

The following section describes the BUSY-OUT screen for the BRM-2 module diagnostics. This screen refreshes periodically.

To display the BUSY-OUT screen for BRM-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a BRM-2 module.

3. Select **Busy Out**. The BUSY-OUT screen for the selected BRM-2 module appears, as shown in **Figure 61**.

```
[Bay 1]                   << B U S Y - O U T >>      't'/'u' - Tag/Untag
                                                     Arrow keys - Change channel
                                                     '.' - To exit screen

Port 1
  Channel        |
  Number         | 1 2
  Status         | + +
  Busy-out Tag   |

Port 2
  Channel        |
  Number         | 1 2
  Status         | B B
  Busy-out Tag   | * *

  Instructions:
  - Use arrow keys to traverse between ports and channels
  - Press 't' to tag a channel or port for busy-out
  - Press 'u' to untag a channel or port for busy-out

  Legends: B - Busied out                 + - Idle
           i - Busying out incoming call  X - Offline
           o - Busying out outgoing call  - - Unequipped
           I - Incoming call
           O - Outgoing call              * - Tagged for busy out
```

The Busy Out screen for BRM-2 module diagnostics shows the following elements:

- **Port 1** is highlighted and flashing upon entry to this screen.

- The **Channel Number** identifies the channels available with the trunk programmed for the module port.

- The **Instructions** summarize how to use the screen commands to busy-out channels or ports.

- The **Status** line identifies the status of the channel above it.

- The **Busy-out Tag** line identifies the Busy Out condition of the channel above it.

- The **Legends** area defines the Status line and Busy-out Tag line symbols.

Instead of the navigation guidelines described in "Navigating Through the Digital Trunk Diagnostics Screens" on , the following controls apply:

- Pressing the UP ARROW or DOWN ARROW switches the selection between ports.

- From a port selection, pressing the LEFT ARROW moves the cursor to Channel 2 in that port.

- From a port selection, pressing the RIGHT ARROW moves the cursor to Channel 1 in that port.

- Pressing the LEFT ARROW or RIGHT ARROW moves the cursor between channels in the same port.

- Pressing the LEFT ARROW key from Channel 1moves the cursor to Channel 2 in the same port.

- Pressing the RIGHT ARROW from the Channel 2 moves the cursor to Channel 1 in the same port.

- Pressing the UP ARROW while a channel is selected selects the channel's port.

- Pressing the DOWN ARROW while a channel is selected select the other port.

- If a port is selected, pressing the T key or the t key tags all channels in that port for busy-out.

- If a port is selected, pressing the U key or the u key untags all channels in that port for busy-out.

- If a channel is selected, pressing the T key or the t key tags that channel for busy-out.

- If a channel is selected, pressing 'the U key or the u key untags that channel for busy-out.

VERSION INFORMATION

The following section describes the Version Information screen for BRM-2 module diagnostics.

To display the Version Information screen for BRM-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a BRM-2 module.

3. Select **Version**. The Version Information screen appears, as shown in **Figure 62**.

```
Version Information                                                 [Bay 1]
==================


BRI Application
  Version:         2.3
  Build Date:      Wed May 23 2007 11:23:58
  Build directory: /home/ymc/it5k_ro_2_0_devel_ws/RemoteOffice/intl/apps/bri

DSP
  Version:         0.0



Press any key to exit
```
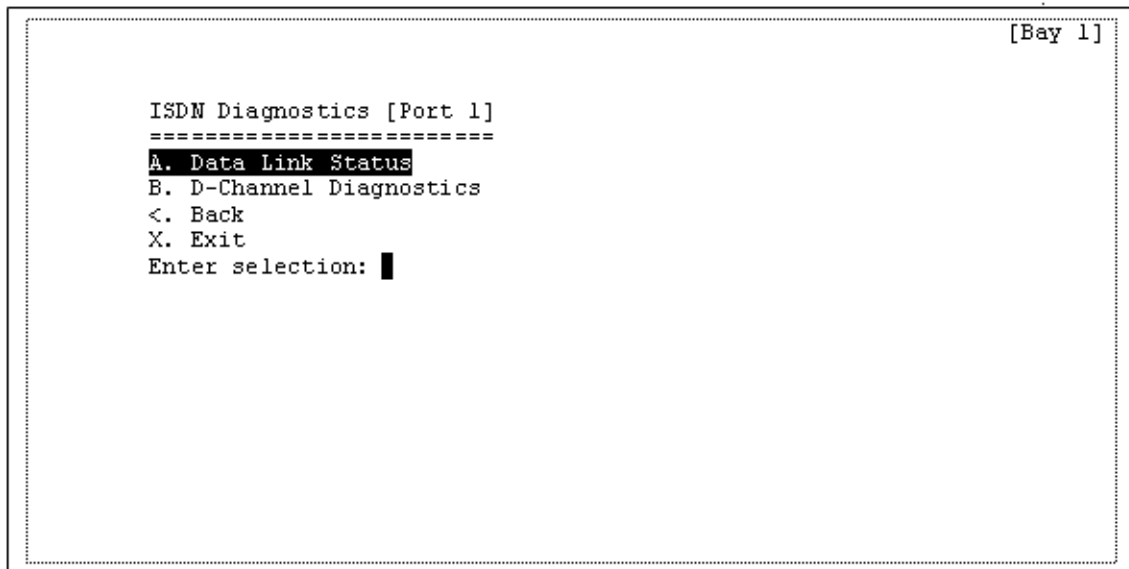
The format of the version numbers varies in accordance with data that is available for each software/hardware module.

ISDN DIAGNOSTICS PORT 1 OR 2

The following section describes the ISDN Diagnostics screen supporting BRM-2 module diagnostics.

To display an ISDN Diagnostics screen for a BRM-2 port:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a BRM-2 module.

3. Select **ISDN Diagnostics Port 1** or **ISDN Diagnostics Port 2**, as applicable. The ISDN Diagnostics screen for the selected port appears, as shown in **Figure 63**.

```
                                                                  [Bay 1]


        ISDN Diagnostics [Port 1]
        =========================
        A. Data Link Status
        B. D-Channel Diagnostics
        <. Back
        X. Exit
        Enter selection: █
```

The **Data Link Status** selection is highlighted upon entry to this screen.

Typing the selection shortcuts—A or a, B or b, <, X or x—activates the corresponding selection.

Pressing **Enter** activates the highlighted selection.

The cursor is located at the **Enter selection** prompt upon entry to this screen. Once the UP ARROW or DOWN ARROW has been pressed, from that time forward, the cursor is located at the end of the highlighted selection.

ISDN DIAGNOSTICS PORT 1 OR 2 – DATA LINK STATUS

The following section describes the port ISDN Data Link Status screen supporting BRM-2 module diagnostics. The Data Link Status screen is useful to troubleshoot OSI model Layer 2 problems on an ISDN connection. This screen refreshes periodically.

To display a port ISDN Data Link Status screen for BRM-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a BRM-2 module.

3. Select **Port**.

4. Select **ISDN Diagnostics Port 1** or **ISDN Diagnostics Port 2**, as applicable.

5. Select **Data Link Status**. The ISDN Data Link Status screen for the module port appears, as shown in **Figure 64**.

```
[Bay 1] Port #1                          SPACE - Next DLCB
                                         '.' - To exit screen
BRI (ISDN) Configuration
Switch Type... National ISDN 1    K.......... 1
T200........ 1000 ms              N200....... 3
T202........ 2000 ms              N201....... 260
T203........ 30000 ms             N202....... 3
T204........ 2000 ms              N204....... 3


Data Link Control Block #1
D-Channel Status... UP
Sapi.............. $00
Tei............... $00
State............. MULTIPLE FRAME ESTABLISHED
VA................ $37
VS................ $37
VR................ $3B
```

If only one Data Link Control Block (DLCB) exists for the current port, pressing the SPACEBAR, displays the next port's Data Link Status screen, if equipped. If more than one exists, pressing the SPACEBAR to display the next DLCB, the Data Link Control Block # changes accordingly.

If the other port is unequipped, the **Any other key - Next Port** option is omitted. The option may appear or disappear if the other port is equipped or unequipped while on this screen.

ISDN DIAGNOSTICS PORT 1 OR 2 – D-CHANNEL DIAGNOSTICS

The following section describes the port D-Channel Diagnostics screen supporting BRM-2 module diagnostics. The D-Channel Diagnostics screen allows you to enable or disable the D-Channel Diagnostics for the current port. This screen refreshes periodically.

To display a port ISDN Data Link Status screen for BRM-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a BRM-2 module.

3. Select **Port**.

4. Select **ISDN Diagnostics Port 1** or **ISDN Diagnostics Port 2**, as applicable.

5. Select **D-Channel Diagnostics**. The D-Channel Diagnostics screen for the module port appears, as shown in **Figure 65**.

```
                                                        [Bay 1]


        D-Channel Diagnostics: Port 1: [DISABLED]
        =======================================
        A. Enable D-Channel Diagnostics (Level-1)
        B. Enable D-Channel Diagnostics (Level-2)
        C. Disable D-Channel Diagnostics
        D. Display D-Channel Diagnostics
        <. Back
        X. Exit
        Enter selection:
```

The **Enable D-Channel Diagnostics (Level-1)** selection is highlighted upon entry to this screen.

For the T1M no reference to Port # appears in the menu title.

The state of the current port's D-Channel Diagnostics is enclosed in brackets and can have one the following values:

- DISABLED

- LEVEL-1

- LEVEL-2

- UNDEFINED

The **Display D-Channel Diagnostics** option has no effect if the D-Channel Diagnostics is disabled.

ISDN DIAGNOSTICS PORT 1 OR 2 – D-CHANNEL DIAGNOSTICS – DISPLAY

The following section describes the port D-Channel Diagnostics Display screen supporting BRM-2 module diagnostics. The D-Channel Diagnostics Display screen allows you to view the D-Channel Diagnostics in real-time.

To display a port D-Channel Diagnostics Display screen for BRM-2 diagnostics:

1. From the DMU Main Menu select **Digital Trunk Diagnostics**.

2. Select a **Bay** containing a BRM-2 module.

3. Select **Port**.

4. Select **ISDN Diagnostics Port 1** or **ISDN Diagnostics Port 2**, as applicable.

5. Select **D-Channel Diagnostics**.

6. Select **Enable D-Channel Diagnostics Level-1** or **Level-2**, as applicable.

7. Select **Display D-Channel Diagnostics**, The D-Channel Diagnostics Display screen for the module port appears, as shown in **Figure 66**.

```
                                                                    [Bay 1]
D-Channel Diagnostics: Port 1: [LEVEL-1]
========================================

Pressing any key now will start the D-channel Diagnostics real-time display on
the screen.  Once the D-Channel Diagnostics real-time display has started,
pressing the '.' key will exit back to D-Channel Diagnostics menu.

Press any key to continue: █
```

The state of the D-Channel diagnostics for the current port is displayed in brackets. Possible values include:

- LEVEL-1
- LEVEL-2
- UNDEFINED

DB TEST AND REPAIR UTILITY

The DB Test and Repair utility tests databases for corruption and referential integrity. For more information, refer to the DB Test and Repair Help.

BUSY OUT MANAGER

The Busy Out Manager allows you to select ports or circuits from a list and specify Busy Out commands for the selected items. The Busy Out Manager displays all ports on the selected module as well as what type of port is configured (a "None" device will be shown if the port has not been configured). Selecting a port displays the devices/circuits programmed for that particular port (devices/circuits that have not been configured will not show up as a None device). See "Programming the Busy Out Manager" on **page 213** for details.

DATABASE CHANGE LOG

This section contains the following information:

- **General Guidelines** below

- How to Read the Database Change Log on **page 18-1099**

- Troubleshooting the Database Change Log on **page 18-1112**

The Database Change log provides details on user changes to DB Programming. The log details recent user changes to facilitate diagnostics. The Database Change log is written and maintained by the MiVoice Office 250. It is stored on the MiVoice Office 250 and is accessible through the Administrative Web Portal (AWP) interface. For more information, refer to AWP Help.

The naming convention for the log filename is cp_database_log_<*date and time*>.txt, where date and time indicates when the file was created. The maximum file size allowed is 200 KB. Like other system logs, after the first log file reaches the maximum size, the system creates a new log file. After the second log file reaches the maximum size, the system deletes the oldest backup log file, and then logs to a new log file. The MiVoice Office 250 maintains two backup log files.

GENERAL GUIDELINES

The following list provides some general information about the Database Change log:

- All entries include the path to the folder or menu option from which the change was made, the name of the field that was changed, and the new value (if applicable).

- Numbers are translated to text whenever possible.

- The Database Change log file does not include DB Programming changes that occur behind the scenes. Some examples are:

  - You put a phone in Class of Service 2. Behind the scenes, DB Programming assigns the phone to User Group 1.

  - You delete a SIP Voice Mail. Behind the scenes, DB Programming deletes all mailboxes, group lists, and applications under that SIP Voice Mail.

- You unequip phone 1003, which was the attendant for phone 1004. Behind the scenes, DB Programming sets the attendant for 1004 to NONE.

- You delete a trunk group. Behind the scenes, DB Programming puts all trunks that were in that group into the unused trunk group. Then DB Programming goes through and replaces all links to that trunk group with the appropriate replacement (usually NONE).

- Changes are included for Remote sessions only. Local session changes are not included, other than the indication of when you perform a Database Restore. See **page 1107** for an example.

- Changes made to DB Programming outside of the DB Programming application are not included in this log. Examples of changes that are not included:

  - Administrator phone programming

  - Automatic exports from other nodes

  - Equip off-node device event

  - Automatic DB Backup Save/Restore attempts

  - System OAI

  - User phone programming

  - Date/Time updates

HOW TO READ THE DATABASE CHANGE LOG

Database Change Log entries are categorized as follows:

- **General** below

- **Header and Footer** below

- Field Changes on **page 18-1100**

- Other Changes on **page 18-1102**

- Tools Menu on **page 18-1104**

- Operations Menu on **page 18-1106**

- View Menu on **page 18-1111**

GENERAL

This section details some general information about the Database Change Log:

- Every log entry begins with the date and military time:

```
    [2008-01-30 11:20:11 DBP]   <text>
```

- The "DBP" service name that is included after the timestamp does not appear if the log file is stored on the DB Programming computer (instead of the MiVoice Office 250).

- All changes are logged regardless if they are in On-Line Monitor (OLM) mode or not. OLM-only field changes are designated as "OLM."

```
[YYYY-MM-DD-HH:MM:SS DBP] [Folder] Changed <Field> to <New Value> - OLM
```

HEADER AND FOOTER

This section contains information about the header and footer of the Database Change Log.

- The following text appears for each session that is initiated:

```
[2008-01-30 11:19:48 DBP] CP SESSION ESTABLISHED - Database
Programming Client Address: 192.168.1.37
[2008-01-30 11:20:11 DBP] User Information: jsmith on smith2-0xp
[2008-01-30 11:20:11 DBP] Database Programming Version: 3.0.1.5
[2008-01-30 11:20:12 DBP] Call Processing Version: 3.0.1.13
[2008-01-30 11:20:12 DBP] Voice Processor Type: Basic
[2008-01-30 11:20:12 DBP] Voice Processor Version: 3.0.1.13
```

The version numbers for each component clearly show when you perform an upgrade. You can also search by a specific component version.

- The following text appears for each session that is terminated:

```
[2008-01-30 11:15:27 DBP] SESSION TERMINATING WITH <status>
[2008-01-30 11:15:27 DBP] CP SESSION TERMINATED - Database
Programming Client Address: 192.168.1.37
```

- The termination status includes the following:

    - **SUCCESS**: The session terminated with success.

    - **WARNING – <text>**: The session completed, but warning text appears.

    - **FAILURE – <text>**: An error caused the session to terminate. Error text appears.

    If the session terminates in such a way that DB Programming does not post a termination entry to the log, only the CP SESSION TERMINATION message appears.

FIELD CHANGES

This section details the field changes in the Database Change Log. Many database changes are made through editing a field. When you edit a control and complete the edit, the change is saved to the database. These changes are always associated with a field in a folder hierarchy.

Each field change includes the following entry:

```
[YYYY-MM-DD-HH:MM:SS DBP] [Folder] Changed <Field> to <New Value>
- OLM
```

Some examples are:

```
[2008-01-28 12:58:22 DBP] [Passwords\2] Set Description to John
[2008-01-28 12:58:35 DBP] [Passwords\2] Set Password to ########
[2008-01-28 12:58:59 DBP] [System Forwarding Paths\2] Set
Description to test2
[2008-01-28 12:59:18 DBP] [System Forwarding Paths\2] Set
Forwarding Point 3 to 1001
[2008-01-30 13:22:23 DBP] [System Information\Meta Database] Set
Echo Profiles to 26 - OLM
```

- Where [*Folder*] is the path to the folder in which the field resides. This is limited to two levels up, except in the following situations:

    - "MiVoice Office 250" appears only for items that are changed at the highest level.

    - "System" appears for only items that are changed at the System level.

    - If the second level up is a number or shows "local," one more level appears.

    - The folder path will always go up high enough to show the affected extension.

- Where <*Field*> is the column header or field name.

- Where <*New Value*> is the new value.

- "OLM" appears for OLM-only fields.

OTHER CHANGES

This section details other changes in DB Programming:

- **Device Additions and Deletions**:
    - The device type and extension(s) always appear.
    - The hardware address appears when it is available.
    - Items added in a batch are grouped on one line only when they do not have hardware addresses. A batch item with a hardware address appears on its own line.
    - Some examples are:

```
[2008-01-30 13:24:18 DBP] [Phones] Added 1001
[2008-01-30 13:24:18 DBP] [1001] Set MAC Address to
00:10:36:00:10:01
[2008-01-30 13:24:43 DBP] [Phones] Added 1002,1003,1004
[2008-01-30 13:24:43 DBP] [1002] Set MAC Address to
00:10:36:02:FF:FF
[2008-01-30 13:24:43 DBP] [1003] Set MAC Address to
00:10:36:03:FF:FF
[2008-01-30 13:24:44 DBP] [1004] Set MAC Address to
00:10:36:04:FF:FF
[2008-01-30 13:25:05 DBP] [Trunks] Added 94000
[2008-01-30 13:25:17 DBP] [Trunks] Added 94001
[2008-01-30 13:25:17 DBP] [94001] Set Phone Name to E1
[2008-01-30 13:25:17 DBP] [94001] Set Gateway Name to G1
[2008-01-30 13:25:48 DBP] [Bay 1: Loop Start Module - 2\01.01]
Added 94002 (Loop Start)
[2008-01-30 13:25:49 DBP] [Bay 1: Loop Start Module - 2\02.01]
Added 94003 (Loop Start)
[2008-01-30 13:26:30 DBP] [01:02.01 T1\T1 Circuits\01.01] Added
1005 (Single Line)
[2008-01-30 13:26:38 DBP] [01:02.01 T1\T1 Circuits\01.02] Added
94004 (Ground Start)
[2008-01-30 13:26:44 DBP] [01:02.01 T1\T1 Circuits\01.03] Added
94005 (DID)
[2008-01-30 13:26:49 DBP] [01:02.01 T1\T1 Circuits\01.04] Added
94006 (E&M)
[2008-01-30 13:27:24 DBP] [01:03.01 E1/PRI\E1 Circuits\01.01]
Added 94007 (B-Channel)
[2008-01-30 13:27:25 DBP] [01:03.01 E1/PRI\E1 Circuits\01.02]
Added 94008 (B-Channel)
[2008-01-30 13:27:27 DBP] [01:03.01 E1/PRI\E1 Circuits\01.03]
Added 94009 (B-Channel)
[2008-01-30 13:27:28 DBP] [01:03.01 E1/PRI\E1 Circuits\01.04]
Added 94010 (B-Channel)
[2008-01-30 13:27:42 DBP] [Hunt Groups] Added 2000
[2008-01-30 13:27:53 DBP] [Page Zones] Added 9600,9601,9602
```

- **List Additions and Deletions and Drag, Drop, and Move**:

  - The item type always appears.

  - The extension appears for lists that have an extension.

  - Added, deleted, or moved items are listed by ID or extension.

  - Some examples are:

  ```
  [2008-01-30 13:27:53 DBP] [Page Zones] Added 9600,9601,9602
  [2008-01-30 13:30:27 DBP] [System Speed Dial] Added
  001,002,003,004
  [2008-01-30 13:31:30 DBP] [Hunt Groups\2000\Members] Added
  1005,10220,10221 starting in position 1
  [2008-01-30 13:32:01 DBP] [Hunt Groups\2000\Supervisors] Added
  1000
  [2008-01-30 13:32:32 DBP] [Forced\Non-Validated\All Calls] Added
  1001,1002
  [2008-01-30 13:32:44 DBP] [Forced\Non-Validated\Long-Distance Toll
  Calls] Added 1002
  [2008-01-30 13:32:59 DBP] [Account Codes\None] Added 1001
  [2008-01-30 13:33:07 DBP] [Forced\Non-Validated\Long-Distance Toll
  Calls] Removed 1002
  [2008-01-30 13:33:37 DBP] [Page Zones] Deleted 9602
  ```

- **Change Extension**:

  - Each extension appears on a separate line even if they are batch mode changes.

  - Some examples are:

  ```
  [2008-01-30 13:34:39 DBP] [Phones\1002] Changed extension to 2001

  [2008-01-30 13:34:39 DBP] [Phones\1003] Changed extension to 2002

  [2008-01-30 13:34:39 DBP] [Phones\1004] Changed extension to 2003
  ```

- **Copy and Paste**:
  - The devices copied from and to are listed along with a list of attributes included in the copy.
  - Some examples are:

```
[2008-01-30 15:45:26 DBP] [Loop Start\94257] Copied to 94258:
        Answer Supervision Type
        Connect Trunk-to-Trunk Call On Polarity Reversal
        DTMF Signaling
        Hybrid Balance
        Language
        Receive Gain
        Send Digits En Bloc
        Service Type
        Transmit Gain
        ---
 [2008-01-30 15:46:09 DBP] [Hunt Group\2000] Copied to 2002:
        ACD Agent No Answer - DND Message Additional Text
        ACD Agent No Answer - DND Message Number
        ACD Hunt Group
        Agents
        Analog Voice Mail Hunt Group
        Announcement
        Audio for Calls Camped onto this Device
        Audio for Calls Ringing this Device
        Audio for Camped-On Announcement Calls
        Camp-Ons Allowed
        Group Call Pick-up
        Members
        Overflow
        Priority Level
        Recall
        Restart ACD Idle Time Upon Login
        Return ACD Calls to Hunt Group
        Search Type
        Send Camp-On Notifications to Members in DND
        Supervisors
        Timers
        Use ACD Agent IDs
```

- **Other Dialogs**:
  - For changes made through dialogs (other than the wizards discussed later in the next section), the context is included in square brackets, and the field name appears with the new value.
  - Some examples are:

```
[2008-01-31 14:09:53] [Key Assignments\IP/Digital Telephone\1]
Changed Key 17 to Secondary Extension Key: 10220 - 5 rings
[2008-01-31 14:09:53] [Key Assignments\IP/Digital Telephone\1]
Changed Key 19 to DSS/BLF Key: 10221
[2008-01-31 14:10:01] [Key Assignments\IP/Digital Telephone\1]
Changed Key 32 to Programmable Key 15
```

TOOLS MENU

This section details the Tools Menu options of the Database Change Log:

- **Configuration Wizard**:

- Consists of a multi-line entry.

- "Configuration Wizard" appears as the first entry and a header.

- A line of text appears for each board and device configured and for each programming change.

- Examples:

```
[2008-02-06 06:28:45 DBP] Configuration Wizard:
        Set SSH Server Enabled to No
        Set Web Server Enabled to Yes
        Set SSH Server Port to 22
        Set Listening Port to 4000
        Set Web Listening Port to 80
        Set PPP IP Address to 192.168.201.209
        [Mitel 3.0.2.5] Added 1001
        [1001] Set MAC Address to 00:10:36:00:10:01
        [Bay 1] Added Loop Start Module - 4
        [Bay 2] Added Dual T1/E1/PRI Module
        [Mitel 3.0.2.5\01.01] Added 94000 (Loop Start)
        [Mitel 3.0.2.5\02.01] Added 94001 (Loop Start)
        [Mitel 3.0.2.5\03.01] Added 94002 (Loop Start)
        [Mitel 3.0.2.5\04.01] Added 94003 (Loop Start)
        [Bay 2 Port 1] Added T1 Module
        [Bay 2 Port 2] Added E1/PRI Module
        [DEI 1 Bay 1] Added Digital Endpoint Module - 16
        [Mitel 3.0.2.5\01.01] Added 1002 (Digital Telephone)
        [Mitel 3.0.2.5\01.02] Added 1003 (Single Line)
```

- **Networking Wizard**:

  - Consists of a multi-line entry.

  - "Networking Wizard" appears as the first entry and a header.

  - A line of text appears for each board and device configured and for each programming change.

  - Examples:

```
[2008-02-06 06:31:20 DBP] Networking Wizard:
        [Mitel 3.0.2.5] Added 97002
        [Mitel 3.0.2.5] Added P8000
        Set Node IP Connection Group to Conn to Node 2
        Set Description to Chandler
        Set Username to CHANDLER
        [Mitel 3.0.2.5] Added P6001
        Set Remote IP Address to 192.168.200.208
        Set Remote Audio Receive Port to 6004
        Set Remote Listening Port to 5570
        Set Description to ChandlerNode
        Set Username to CNODE
        [Remote Node\97002] Added Node Trunk/IP Connection Groups
(P8000) in position 1
```

- Example of results from the T1/PRI Networking Wizard:

```
[2008-02-06 06:33:58 DBP] Networking Wizard:
        Set Node to 168
        [Mitel 3.0.2.5] Added 97501
        Set Description to Conn to Node 2
        Set Username to
        Set Reference Clock List to 2
        [Mitel 3.0.2.5\02.01] Added 94004 (B-Channel)
        [Mitel 3.0.2.5\02.02] Added 94005 (B-Channel)
        [Mitel 3.0.2.5\02.03] Added 94006 (B-Channel)
        [Mitel 3.0.2.5\02.04] Added 94007 (B-Channel)
        [Mitel 3.0.2.5\02.05] Added 94008 (B-Channel)
        [Remote Node\97002] Added Node Trunk/IP Connection Groups
(P8000,97501) starting in position 1
```

- **Resource Reservation Tool**:
  - Consists of a multi-line entry.
  - "Resource Reservation Tool" appears as the first entry and a header.
  - A line of text appears for each programming change.
  - The old value appears along with the new value.
  - Examples:

```
[2008-02-06 06:36:19 DBP] Resource Reservation Tool:
        Set [1000] Reserved to Yes
        Changed G.711 Phones from 0 to 2 - OLM
        Changed G.711 Trunks from 0 to 3 - OLM
        Changed G.729 Phones from 0 to 6 - OLM
        Changed G.729 Networking from 0 to 9 - OLM
        Changed Emergency/911 Resources Reserved from 1 to 2
        Changed Unified Voice Messaging Port Resources Reserved from
2
        Changed Maximum Simultaneous Fax Over IP (T.38) from 0 to
2
```

OPERATIONS MENU

This section details the Operations Menu options of the Database Change Log:

- **Database Save**:
  - "Begin Database Save To <*path*>" appears as the first entry.
  - "Begin Voice Data Save To <*path*>" appears as the second entry if voice data was saved also.
  - The following entry consists of one of the following messages:
    - "Completed with Success" when the operation completes successfully.
    - "Terminated with Warning – <*warning*>" when the operation terminates with a warning.
    - Terminated with Failure – <*error*>" when the operation fails.

- Examples:

```
[2008-02-05 20:03:32 DBP] Begin Database Save To <path>...
[2008-02-05 20:03:32 DBP] Begin Voice Data Save To <path>...
[2008-02-05 20:10:37 DBP] Completed with Success
[2008-02-05 20:10:37 DBP] Terminated with Warning - <warning>
[2008-02-05 20:10:37 DBP] Terminated with Failure - <error>
```

- **Backup Database Save**:
  - "Begin Backup Database Save" appears as the first entry.
  - The following entry consists of one of the following messages:
    - "Completed with Success" when the operation completes successfully.
    - "Terminated with Warning – <*warning*>" when the operation terminates with a warning.
    - Terminated with Failure – <*error*>" when the operation fails.
  - Examples:

```
[2008-02-05 20:10:33 DBP] Begin Backup Database Save...
[2008-02-05 20:10:37 DBP] Completed with Success
[2008-02-05 20:10:37 DBP] Terminated with Warning - <warning>
[2008-02-05 20:10:37 DBP] Terminated with Failure - <error>
```

- **Database Restore**:
  - "Begin Database Restore From <*path*>" appears as the first entry.
  - "Begin Voice Data Restore From <*path*>" appears as the second entry if voice data was restored also.
  - The following third entry consists of one of the following messages:
    - "SESSION TERMINATING WITH SUCCESS" when the operation completes successfully.
    - "SESSION TERMINATING WITH WARNING: <*warning*>" when the operation terminates with a warning.
    - "SESSION TERMINATING WITH ERROR: <*error*>" when the operation fails.
  - "CP SESSION TERMINATED..." is the last entry always because the programming session always terminates after a Database Restore operation.
  - Examples:

```
[2008-02-05 20:03:32 DBP] Begin Database Restore From <path>...
[2008-02-05 20:03:32 DBP] Begin Voice Data Restore From <path>...
[2008-02-05 20:03:48 DBP] SESSION TERMINATING WITH SUCCESS
[2008-02-05 20:03:48 DBP] SESSION TERMINATING WITH WARNING:
<warning>
[2008-02-05 20:03:48 DBP] SESSION TERMINATING WITH FAILURE:
<error>
[2008-02-05 20:03:50 DBP] CP SESSION TERMINATED - Database
Programming Client Address: 192.168.1.37
```

- **Default Database**:
  - "Begin Default Backup Database" appears as the first entry.
  - The next entry consists of one of the following messages:
    - "Completed with Success" when the operation completes successfully.
    - "Terminated with Warning – *<warning>*" when the operation terminates with a warning.
    - Terminated with Failure – *<error>*" when the operation fails.
  - "Default Database" appears as the third entry.
  - The fourth entry consists of one of the following messages:
    - "SESSION TERMINATING WITH SUCCESS" when the operation completes successfully.
    - "SESSION TERMINATING WITH WARNING: *<warning>*" when the operation terminates with a warning.
    - "SESSION TERMINATING WITH ERROR: *<error>*" when the operation fails.
  - "CP SESSION TERMINATED..." is the last entry always because the programming session always terminates after a Database Restore operation.
  - Examples:

```
[2008-02-05 06:05:45 DBP] Begin Default Backup Database...
[2008-02-05 06:05:45 DBP] Completed with Success
[2008-02-05 06:01:40 DBP] Default Database
[2008-02-05 06:01:44 DBP] SESSION TERMINATING WITH SUCCESS
[2008-02-05 06:01:44 DBP] CP SESSION TERMINATED - Database
Programming Client Address: 192.168.1.37
```

- **Error Information**:
  - "Error Information History Queue Frozen" appears as the first entry when a history queue freeze occurs.
  - "Begin Error Information Save" appears as the next entry when the error information is saved.
  - The next entry consists of one of the following messages:
    - "Completed with Success" when the operation completes successfully.
    - "Terminated with Warning – *<warning>*" when the operation terminates with a warning.
    - Terminated with Failure – *<error>*" when the operation fails.
  - "Error Information History Queue Unfrozen" is the last entry when a history queue unfreeze occurs.
  - Examples:

```
[2008-02-06 06:01:35 DBP] Error Information History Queue Frozen
[2008-02-06 06:01:46 DBP] Begin Error Information Save...
[2008-02-06 06:01:49 DBP] Completed with Success
[2008-02-06 06:01:52 DBP] Error Information History Queue Unfrozen
```

- **Export/Import Devices**:
    - "Begin Export" or "Begin Import" appears as the first entry when a device is exported or imported.
    - The nodes are listed next, followed by the device types.
    - After the list of device types, "Results" appears with a timestamp, followed by the specific results for each node.
    - The last entry includes one of the following messages:
        - "Completed with Success" when the operation completes successfully.
        - "Terminated with Warning – *<warning>*" when the operation terminates with a warning.
        - Terminated with Failure – *<error>*" when the operation fails.
    - An example of a successful export is:

```
[2008-02-06 14:33:50 DBP] Begin Export...
        Nodes included:
                1    Robec
        Device Types included:
                Digital Telephone
                Single Line
                Hunt Group
                Page Port
                Page Zone
                Voice Mail
                Message Notification/Retrieval
                Call Routing Announcement
                Auto Attendant Recall
                Auto Attendant
                Record-A-Call
                Scheduled Time-Based Application Router (STAR)
                Unassociated Mailbox Off-Node Device
                Group List Off-Node Device
                B-Channel Station Off-Node Device
                IP Connection
                Modem
                Phantom Device
                Network Group
                ACD Agent ID
[2008-02-06 14:33:51 DBP] Results:
        Node 3: Export Source
        Node 1: COMPLETED
[2008-02-06 14:33:51 DBP] Completed with Success
```

    - An example of a failed export is:

```
[2008-02-06 15:00:55 DBP] Begin Export...
        Nodes included:
                1    Robec
        Device Types included:
                Single Line
[2008-02-06 15:01:11 DBP] Results:
        Node 3: Export Source
        Node 1: Err -- Not Reachable
[2008-02-06 15:01:14 DBP] Terminated with Failure - The operation
failed for one or more nodes.
```

- **IP Device Status**: There is no logging information for this operation.

- **Print Reports**: There is no logging information for this operation.

- **Upload Software License**:

    - "Begin Upload Software License" appears as the first entry along with the list of differences between the current license and the new license that will be uploaded.

    - The second entry includes one of the following messages:

        - "Completed with Success" when the operation completes successfully.

        - "Terminated with Warning – *<warning>*" when the operation terminates with a warning.

        - Terminated with Failure – *<error>*" when the operation fails.

    - The third entry appears if the license update requires a reset. This entry consists of one of the following messages:

        - "SESSION TERMINATING WITH SUCCESS" when the operation completes successfully.

        - "SESSION TERMINATING WITH WARNING: *<warning>*" when the operation terminates with a warning.

        - "SESSION TERMINATING WITH ERROR: *<error>*" when the operation fails.

    - "CP SESSION TERMINATED..." appears as the last entry if the license update requires a reset because the session always terminates after a reset.

    - Examples:

    ```
    [2008-02-06 06:11:00 DBP] Begin Upload Software License...
            Changed Creation Date from Feb 01, 2008 (13:51:28) to Feb
    04, 2008 (12:47:13)

            Changed Digital Expansion Interface # 3 from No to Yes
            Changed File-Based MOH Sources from 3 to 4
    [2008-02-06 06:11:03 DBP] Completed with Success
    [2008-02-06 06:11:03 DBP] SESSION TERMINATING WITH SUCCESS
    [2008-02-06 06:11:04 DBP] CP SESSION TERMINATED - Database
    Programming Client Address: 172.30.203.19

        If a reset was required.
    ```

- **System Manager CA Certificate Upload**:

    - "Upload System Manager CA Certificate from *<path>*" appears as the first entry.

    - The second entry includes one of the following messages:

        - "Completed with Success" when the operation completes successfully.

        - "Terminated with Warning – *<warning>*" when the operation terminates with a warning.

        - Terminated with Failure – *<error>*" when the operation fails.

    - Examples:

    ```
    [2008-02-06 06:11:00 DBP] Upload System Manager CA Certificate
    from <path>
    [2008-02-06 06:11:03 DBP] Completed with Success
    ```

- **Voice Processor Save or Restore**:
    - "Voice Processor [Save/Restore] (<*type*>) [to|from] <*path*>" appears as the first entry (where <type> is the radio button selected, for example Audiotex Recordings, Group Lists, Mailbox Information, and so forth).
    - The second entry appears with the details about the saved or restored type of information. The example below includes Mailbox Information.
    - The second entry includes one of the following messages:
        - "Completed with Success" when the operation completes successfully.
        - "Terminated with Warning – <*warning*>" when the operation terminates with a warning.
        - Terminated with Failure – <*error*>" when the operation fails.
    - An example of a Database Save for mailbox information is:

```
[2008-07-11 17:15:24 DBP] Begin Voice Processor Save Mailbox Info
To U:/...
        Mailbox Info Included: Name/Greeting,Saved Msgs,New
Msgs,Data
        Items Included: 1000-1001,10220-10221
[2008-07-11 17:15:43 DBP] Completed with Success
```

- **Enable and Disable Unified Voice Messaging**:

    An example of the entry is:

```
[2008-02-06 06:11:00 DBP]  [Enable|Disable] Unified Voice Messaging
```

- **Reset Call Processing Application**:

    An example of the entry is:

```
[2008-02-06 06:11:00 DBP]  Reset Call Processing App
```

- **Reset System**:

    An example of the entry is:

```
[2008-02-06 06:11:00 DBP] Reset System
```

VIEW MENU

This section details the On-Line Monitor option, available from the View Menu, in the Database Change Log. An example of the entry is:

```
[2008-02-06 06:11:00 DBP]  [Enable|Disable] View On-Line Monitor
```

TROUBLESHOOTING THE DATABASE CHANGE LOG

Table 169 lists troubleshooting information for the Database Change Log feature.

Table 169:   Database Change Log Troubleshooting Tips

SYMPTOM	POSSIBLE CAUSE	CORRECTIVE ACTION
Changes are not being recorded in a log on the MiVoice Office 250.	The mode is Local.	You must use a Network connection for database changes to be logged in the MiVoice Office 250.
The Database Change Log and backups do not hold enough information.	The files are too small.	Contact Technical Support to increase the maximum file size in System\Maintenance\Logging Options\Database Change Log in On-Line Monitor mode (OLM). **Note:** Do not use OLM mode unless you are instructed to do so by support personnel.
	There are not enough files.	Contact Technical Support to increase the maximum number of files in System\Maintenance\Logging Options\Database Change Log in On-Line Monitor mode (OLM). **Note:** Do not use OLM mode unless you are instructed to do so by support personnel.

AUDIO DIAGNOSTICS

As an end-user diagnostic tool, the Audio Diagnostics feature allows a user to generate diagnostics information about audio problems. Once the Audio Diagnostics feature is initiated, users are prompted to answer questions about the audio problems by pressing the associated buttons on their phones. Based on the user's selections, the system generates alarm 128, which is displayed on the Administrator phone and on the LCD panel of the chassis. If Automatic Diagnostics Delivery (ADD) is enabled, the collected data is then sent to Mitel Technical Support. By default, ADD is not enabled.

The Audio Diagnostics feature can be accessed when the phone is idle or when the user is on an active call. The amount of diagnostics information that the phone provides to the system depends on the state of the phone when the feature is accessed. For example, an active call produces more diagnostic information than a phone in an idle state. If users do not want to access the Audio Diagnostics feature while on a call, they can access the feature immediately after they hang up.

NOTE	The Audio Diagnostics feature is not available with System OAI Display Control because the external applications have control of the menu buttons and dialpad.

If the user accesses the Audio Diagnostics feature but does not respond to the prompts on the display, the feature times out after 30 seconds. If the user or the other party terminates the call before completing the diagnostics, the feature is terminated. When the feature times out or is terminated, the diagnostics information is not captured.

See **page 419** to enable the Audio Diagnostics flag for a phone.

The Audio Diagnostics folder contains 12 flags that identify possible audio problems. You can have only four of the 12 problem numbers selectable from six-line display phones at any one time. The first four audio problem numbers are selected by default, including:

- 01 – Echo
- 02 – Static
- 03 – One-way audio
- 04 – No audio

This section contains the following information:

- Audio Direction on **page 18-1115**
- Record-A-Call on **page 18-1115**
- Data Collection on **page 18-1116**
- Audio Connections on **page 18-1116**
- Using the Audio Diagnostics Feature on **page 18-1117**
- Responding to the Audio Diagnostics Feature on **page 18-1118**

To enable an Audio Diagnostics problem number:

1. Launch System Administration & Diagnostics, and then start a DB Programming session.

2. From the left pane of Mitel DB Programming, select System – Phone-Related Information – **Audio Diagnostics**. The Audio Diagnostic flags and Yes/No options appear in the right pane, as shown below.
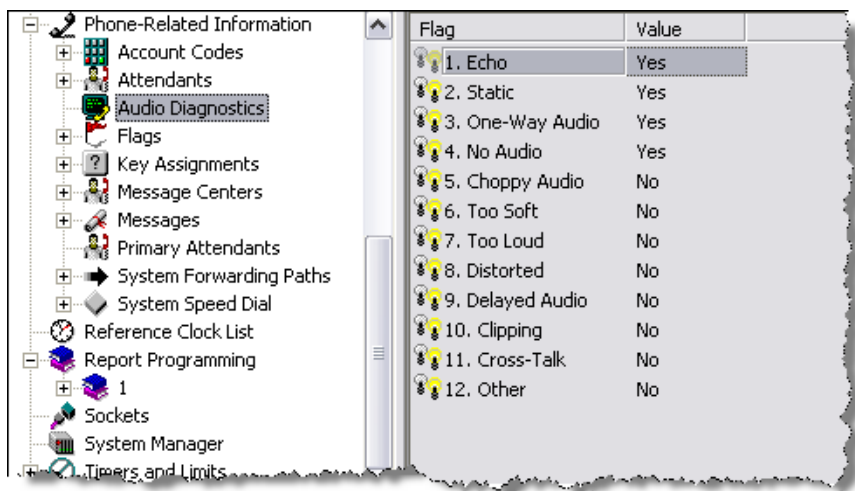
3. Click one of the Yes values to turn it to No.

4. Click the value No value of the flag that you want to enable. The value turns to Yes.

The four selected audio problems appear as menu options on six-line display phones when the user accesses the Audio Diagnostics feature. Six-line users can also enter any of the 12 two-digit problem numbers from their dialpads, but they have the advantage of simply pressing a menu button if the audio problem is one of those enabled in DB Programming.

After accessing the Audio Diagnostics feature, two-line display and non-display phone users have only the option of entering one of the two-digit problem numbers to identify an audio problem. To use the feature, these users would need to have a list of the problem numbers and their meanings. Phantom devices and modems cannot apply the Audio Diagnostics feature code.Audio Problem

When the Audio Diagnostics feature—default feature code 320—is used, the system prompts the user to select a possible audio problem. Six-line display phone users can either press the menu button that corresponds to the displayed audio problem or enter the numeric codes. Users of 2-line display and non-display phones must enter the numeric codes because they do not have menu buttons.

Table 170 shows the 12 audio problems and their associated two-digit codes. However, only four can be enabled in DB Programming. By default, Audio Diagnostics problem numbers 01–04—Echo, Static, One-Way Audio, and No Audio, respectively—are enabled by default for display on six-line display phones.

Table 170: Audio Problem Codes

AUDIO PROBLEM	NUMBER	AUDIO PROBLEM	NUMBER
Echo	01	Too Loud	07
Static	02	Distorted	08
One-Way Audio	03	Delayed Audio	09

Table 170: Audio Problem Codes (continued)

AUDIO PROBLEM	NUMBER	AUDIO PROBLEM	NUMBER
No Audio	04	Clipping	10
Choppy Audio	05	Cross-Talk	11
Too Soft	06	Other	12

Audio Diagnostics: In DB Programming, 12 flags identify audio problems that can be reported using Audio Diagnostics, default feature code 320. Any of the 12 flags can be entered from any phone dialpad after accessing the Audio Diagnostics feature. However, you can have only four problem numbers enabled at any one time to appear on six-line display phones.

In the default state, the first four audio problem flags are enabled, and the remaining eight are disabled. Once the selected audio problems are enabled (set to Yes), the problems are displayed as options on the phone when the user accesses feature code 320. Each audio problem has a unique number (01–12). These numbers are assigned for non-display users to access the feature. For instructions on how to use the feature, see "Using the Audio Diagnostics Feature" on **page 1117**.

Enable Audio Diagnostics: There is a Audio Diagnostics flag under System\Devices and Feature Codes\Phones\<*Node*>\<*Extension*>\Flags that enables the Audio Diagnostics feature for a phone. By default, this flag is disabled. To enable the Audio Diagnostics feature for the phone, set this flag to **Yes**.

Audio Diagnostic Alarm Suppression: This flag programmed under System\Flags gives you the option of suppressing Alarm 128, which is generated when the Audio Diagnostics feature is used. By default, this flag is set to No, which means that the alarm is generated and displayed on the administrator's phone when a user access the Audio Diagnostics feature. If this flag is set to **Yes**, the alarm is suppressed.

AUDIO DIRECTION

After selecting the audio problem, the system prompts the user to choose the direction of the audio problem. Options include:

- Only I hear it (2-line display and non-display users press **1**)

- Only the outside (2-line display and non-display users press **2**)

- We both hear it (2-line display and non-display users press **3**)

RECORD-A-CALL

If the Record-A-Call feature has been programmed for the station, the user can record the call while using the Audio Diagnostics feature.

NOTE	The Record-A-Call feature cannot be used on certain calls such as Agent Help, Station Monitor, Paging, and so forth.

DATA COLLECTION

To collect the diagnostics data, retrieve the Freeze information or send the data to Technical Support via ADD. The diagnostics data that is collected consists of:

- Source phone (extension and module number)

- Destination phone (extension and module number)

- Phone number

- Echo canceller settings (IP Resource)

- Resource manager dump

- Crosspoint/voice channel information

- Current volume levels (near end)

- Hybrid balance values

- De-coupling values

- Network group

AUDIO CONNECTIONS

The system monitors the following audio connections:

- **Audio Connection Status**: The system monitors the connectivity of all IPR application resources configured for private networking to avoid connecting to an IPR application resource that is not connected to the network. To avoid network congestion, the system may take up to 30 seconds to identify an IPR application resource that has lost its connection. During this time, the system may fail to make audio connections for new calls.

- **Audio Connection Parameters**: The system uses the audio connection parameters specified by the node on which the call originates.

- **Audio Route Optimization**: The system attempts to find the audio route that requires the fewest conversions from IP to Pulse Code Modulation (PCM). It can only optimize the audio route between IPR application resources. Also, Audio Route Optimization occurs only when a party, such as a hunt group, multi-ring device, or transferring party merges out of a call. It does not occur during call setup.

- **Insufficient Bandwidth**: The system monitors the in time audio packet thresholds, which are determined by the Average In Time Percentage Threshold and Average In Time Frame Timer fields (see **page 634**). When the system detects that these thresholds have been violated on a call using a vocoder other than G.729B, the system:

    - Displays the transient message, INSUF BANDWIDTH FOR VOICE.

    - Generates a message print warning.

    - Leaves the call active.

    If this occurs, you have the choice of continuing the call or terminating it.

USING THE AUDIO DIAGNOSTICS FEATURE

Only stations with the Audio **Diagnostics station flag** enabled can use the Audio Diagnostics feature. Only 6-line display phones have the menu buttons mentioned in the following instructions. The 2-line display and non-display phones must use the numeric entries to use the Audio Diagnostics feature.

To use the Audio Diagnostics feature:

1. **If the phone is idle,** while on hook enter the Audio Diagnostics feature code (320)

    If you are on an active call, press ∞ then enter the Audio Diagnostics feature code (320).

    The display shows ENTER TWO DIGIT AUDIO PROBLEM.

```
        ENTER TWO DIGIT
         AUDIO PROBLEM
    ECHO
    STATIC
    DELAYED AUDIO
    OTHER
```

2. Do one of the following actions to select an audio problem:

    • Press the desired menu button.

    • Or, enter the two-digit numeric code (01–12) that corresponds to the audio problem—see **page 1114**. Consult your system administrator for the list of Audio Diagnostics codes that are used on your system.

    The display shows PLEASE SELECT THE AUDIO DIRECTION.

3. Select the direction of the audio problem. Do one of the following actions:

    • Press the desired menu button.

    • Press the dialpad button that corresponds to one of the following responses:

        • Press **1** for ONLY I HEAR IT.

        • Press **2** for ONLY OTHER PARTY.

        • Press **3** for WE BOTH HEAR IT.

```
        PLEASE SELECT
       AUDIO DIRECTION

    ONLY I HEAR IT
    ONLY OTHER PARTY
    WE BOTH HEAR IT
```

    If the Record-A-Call feature is enabled for your phone, the display shows WOULD YOU LIKE TO RECORD CALL?

4. If the Record-A-Call feature is available, you have the option of recording the call. Do one of the following actions:

a. Press the desired menu button.

b. Or, press the dialpad button that corresponds to one

```
       WOULD YOU LIKE
       TO RECORD CALL?

    YES
    NO
```
o
f
t
h
e

following responses:

- Press **1** to record the call.
- Press **2** to continue without recording.

  If the system cannot accurately record the call (for example, because different cross-point connections are used) the display shows CALL CANNOT BE COMPLETED.

5. Hang up to complete the Audio Diagnostics feature. System Alarm 128 appears on the system administrator's display phone.

RESPONDING TO THE AUDIO DIAGNOSTICS FEATURE

Once the user completes the Audio Diagnostics feature, the system generates Alarm 128 and the administrator station displays SYS ALARM #128 <EXT> AUDIO FRZ. This alarm indicates the extension that generated the Audio Diagnostics feature.

```
      SYS ALARM #128
      <EXT> AUDIO FRZ
 CLEAR ALARM
 CLEAR ALL ALARMS

 IDLE MENU
```

To respond to Alarm 128:

Alarm 128 indicates that someone has completed the Audio Diagnostics feature, and you need to collect the freeze that contains Message Print entry. Review the freeze diagnostics data. If you need further assistance, submit the data to Technical Support for analysis.

For more information about collecting diagnostics data, contact Mitel Technical Support.

NETWORK GROUP DIAGNOSTICS

Network Group Diagnostics feature allows you to verify that the Network Groups on the local node are programmed correctly. When the Network Group Diagnostics feature code is entered, the system initiates pings from each Mitel IP device on the local node and determines if the other devices respond to the ping. If a device does not respond to the ping or if a firewall is detected, the system issues a Message Print message.

When using Network Group diagnostics, remember the following guidelines:

- The Network Group Diagnostics feature works on Mitel proprietary IP devices for the local node only. It does *not* work across nodes.

- Mitel recommends that you enable a port for Message Print before you run this diagnostics feature.

- Because this feature affects system performance, it is recommended that you run the diagnostics program when the system is idle (that is, after normal business hours).

To run the Network Group Diagnostics:

1. At an administrator phone, enter the Diagnostics On/Off feature code (9900 by default) followed by the Network Group Diagnostics feature code (9963 by default). The display shows NET GRP CHECK (YES = 1, NO = 2).

2. When prompted, press 1 or **ACCEPT** to start the diagnostics feature. The display shows NET GROUP DIAG IN PROGRESS. In Europe, the default Diagnostics On/Off feature code is 9100, and the Network Group Diagnostics feature code is 9163.

When the diagnostics is complete, the phone displays one of the following messages:

- NET GROUP CHECK COMPLETED: Indicates that all IP devices within the Network Groups are capable of communicating via P2P media.

- NET GROUP CHECK ERRORS FOUND: Indicates that either some of the IP devices are offline or there are NATs/firewalls located between the devices. Check Message Print to determine which errors occurred.

OVERSUBSCRIPTION/IP RESOURCE SHARING STATISTICS

The system performs periodic checks to verify whether the IP Device Resource Manager (IPDRM)[1] and the DSP Resource Manager are in sync with IP calls in progress. From the data generated by these checks, the system computes and stores IP resource utilization statistics for diagnostic purposes, including the following measures:

- Average IP resource usage

- Average IP resource Camp On time

- Peak IP resource usage

- Number of times resources are *not* available for a user of a particular vocoder type

---

1. The IP Device Resource Manager maintains all IP resource requests in the system along with all allocated IP resources in the system. It provides useful diagnostics to determine what types of calls are going on in the system and how long calls have been both camped-on or on a call. To dumps the IP Device Resource Manager information, see **page 1139**.

- Number of times resources are *not* available for a user of a particular call type

- Maximum Camp On time

All of the values can be viewed by using either AWP or System Monitor by dumping the IP Resource Statistics or the IPDRM Statistics. Additional information can be collected by dumping the IPDRM or the DSP Resource Manager.

IP RESOURCE SHARING LOG FILE

The IP Resource Sharing log file captures IP resource sharing, or oversubscription, information. You can view this log file from the Log Files page of the Administrative Web Portal (refer to the AWP Help for details).The naming convention for the log filename is cp_sra_log_<date and time>.txt, where date and time indicates when the message was logged to the file. The maximum file size allowed is 200 KB. After the first log file reaches the maximum size, all new messages are rolled over to a second log file.

| NOTE | The system stores only two oversubscription log files at any one time. After the second file is filled, the system wraps messages, overwriting the first log file. |

HYBRID BALANCE TEST

The Hybrid Balance Test automatically measures and assigns the best hybrid balance setting for analog loop start trunks. For optimum audio quality and performance, run the test for all loop-start trunks. Many system capabilities rely on proper Hybrid Balance settings, including echo cancellation (ECAN) and dual tone multi-frequency (DTMF).

You can use Message Print to confirm that the Hybrid Balance Test real-time test results. See "Viewing Hybrid Balance Test Results in Message Print" on **page 1124**.

NOTE	To obtain and maintain optimal Hybrid Balance Settings, do the following: • Perform the test on each LSM port at initial installation of the system. • Perform the test whenever an analog loop start trunk connection is added or changed. • Run the test on a system that is idle or nearly idle. Run the test until the greatest Echo Return Loss (ERL) measurement is within ±2 dB twice in a row, as shown in Expanded Message Print. If the trunk-LSM port interface is optimally balanced, successive ERL readings should not vary much.

This section contains the following information:

- **Improved Hybrid Balance Line Settings** below

- Hybrid Balance Test Options on **page 18-1122**

- Running a Hybrid Balance Test on **page 18-1122**

- Viewing Hybrid Balance Test Results in Message Print on **page 18-1124**

- Viewing Hybrid Balance Results on **page 18-1124**

- Manually Changing the Hybrid Balance Setting on **page 18-1125**

IMPROVED HYBRID BALANCE LINE SETTINGS

There are eight line settings for the Electronic Industries Alliance (EIA) standard loop lengths, as shown in **Table 171**. In DB Programming, you can view the line setting (see Figure 67 on **page 18-1125**) that the system automatically assigns *after* running the Hybrid Balance test. However, you must restart DB Programming to view the setting (see "Running a Hybrid Balance Test" on **page 1122**). You can also manually change this setting in DB Programming (see "Manually Changing the Hybrid Balance Setting" on **page 1125**.

Table 171:   EIA Standard Loop Length Line Settings and Descriptions

EIA LINE	DESCRIPTION
0	Co-located, same AC impedance
1	2000 ft. (609.6 m)
2	7000 ft. (2.14 km)
3	8500 ft. (7.77 km)
4	12,000 ft. (3.66 km)
5	16,500 ft. (5.03 km)
6	30,000 ft. (9.1 km) loaded loop A

Table 171:   EIA Standard Loop Length Line Settings and Descriptions

EIA LINE	DESCRIPTION
7	30,000 ft. (9.1 km) loaded loop B

HYBRID BALANCE TEST OPTIONS

The following are Hybrid Balance Test fields in DB Programming:

- **Hybrid Balance**: The line setting currently assigned to the trunk (see **page 1125**).

- **Measured Echo Return Loss (ERL)**: Shows the optimal ERL value measured during the last hybrid balance test. This field is undefined if the "Last Hybrid Balance Test Timestamp" shows "Untested." Note this field is only updated when DB is first launched.

- **Last Hybrid Balance Timestamp**: Shows the month, day, year, and time of day of the last completed test. If this field contains the string "Untested," the test has never run on the selected trunk and the "Measured Echo Return Loss" field is undefined.

  The Measured Echo Return Loss (ERL) and the Last Hybrid Balance Test Timestamp values are updated only with the start of each DB Programming session. If a test is run during an active session of DB Programming, the fields will *not* be updated with the latest test results until DB Programming is restarted. If Expanded Message Print is enabled, the test results appear in Message Print output. See "Viewing Hybrid Balance Test Results in Message Print" on **page 1124**.

- **Connected to CO**: Indicates whether or not the trunk is connected to another PBX or the CO (see **page 241**).

- **Loop Start AC Impedance**: Indicates the type of line provided by the CO (see **page 267**).

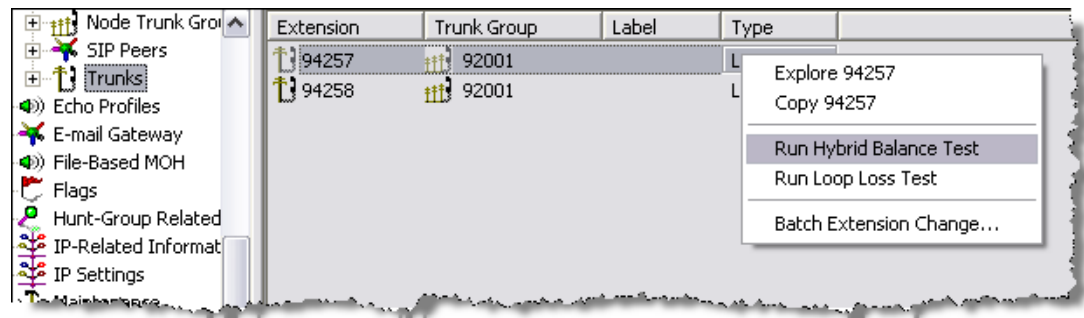## RUNNING A HYBRID BALANCE TEST

You can run a Hybrid Balance Test on a single trunk or on all trunks, as described in the following sections.

RUNNING A HYBRID BALANCE TEST ON SINGLE TRUNK

To run the Hybrid Balance Test on a single trunk:

NOTE	Unless you use Message Print, you must exit and restart DB Programming to view Hybrid Balance Test Results (see "Viewing Hybrid Balance Test Results in Message Print" on **page 1124**).

1. Select System – Devices and Feature Codes – **Trunks**. A list of trunks appears in the right pane, identified by Extension, CO Trunk Group, and Type, as shown below.

2. In the Type column, right-click the individual loop start trunk that you want to test and then click **Run Hybrid Balance Test**. While the Hybrid Balance Test runs, the selected trunk is placed in a Busy condition.

3. If the Hybrid Balance "Help" message appears, read the message, and then click **OK** to continue.

RUNNING A HYBRID BALANCE TEST FOR ALL TRUNKS

To run the Hybrid Balance Test on all trunks:

1. Select System – Devices and Feature Codes – **Trunks**, as shown below.
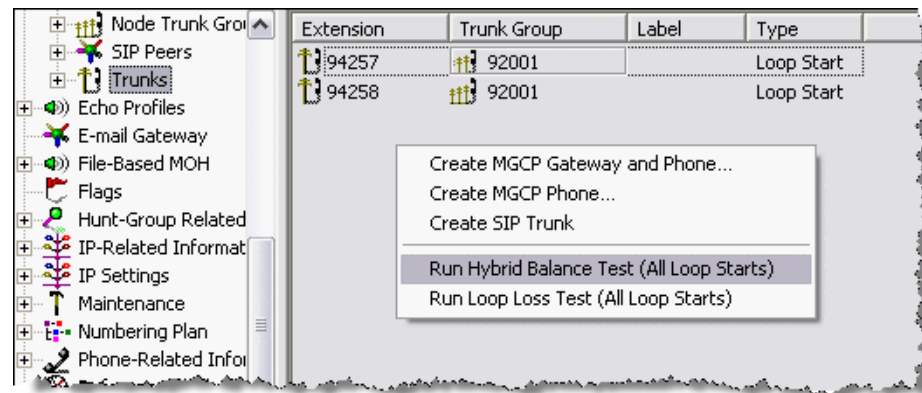
2. Right-click in the right pane, and then select **Run Hybrid Balance Test (All Loop Starts)**. The trunks are made busy during the test and test results for each trunk appears in Message Print. Each set of ERL values and the recommended Optimal Hybrid Balance Setting for each trunk must be processed individually.

3. If the Hybrid Balance "Help" message appears, read the message, and then click **OK** to continue.

# VIEWING HYBRID BALANCE TEST RESULTS IN MESSAGE PRINT

> **NOTICE**
>
> **On-Line Monitor (OLM) Authorization:** To perform the following procedures *only*, you may enter the OLM mode without the supervision of Mitel Technical Support personnel. As required in the procedures, you must make sure that Expanded Message Print is enabled. Any further or other programming in OLM mode is strictly prohibited without specific guidance from Mitel Technical Support personnel. Your cooperation is appreciated.

You can use Message Print in System Administration & Diagnostics to view real-time Hybrid Balance Test results. You can use the results to confirm that the test has run or to manually change the Hybrid Balance Test line setting. For details about Message Print, refer to the *System Administration & Diagnostics Help*.

To set up Expanded Message Print capability:

1. *Before you run a Hybrid Balance Test* (see **page 1122**), on an idle or nearly idle system, launch System Administration & Diagnostics, and then start a DB Programming session.

2. From Mitel DB Programming, select **View**, and then select **On-line Monitor**. The OLM warning message appears.

3. Click **OK**. The OLM view of DB Programming appears, as indicated by a darker background in the right pane and the appearance of previously hidden folders in the left pane.

4. From the left pane, select System – Maintenance – Message Print – **Print Expanded Message Print**. Make sure the value is set to **Yes**. The default is No.
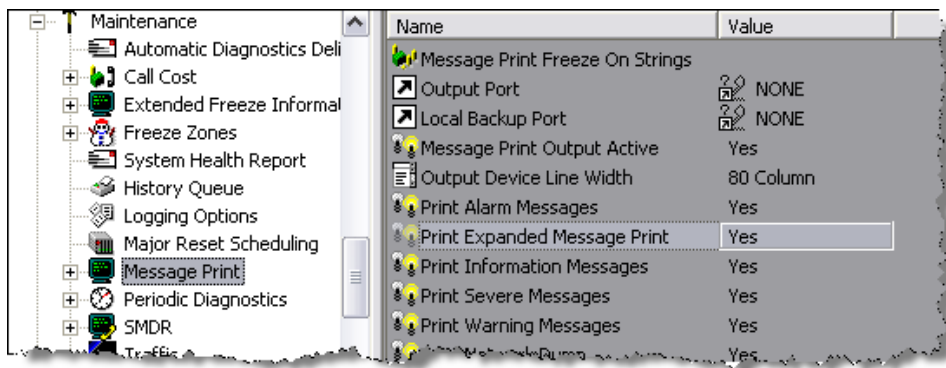
5. After verifying that the Print Expanded Message Print option is enabled, click View and then click online Monitor to clear the option. You exit OLM mode and the regular DB Programming window appears. As applicable, proceed either to the procedure to run a Hybrid Balance Test on a single trunk or on all trunks. See "Running a Hybrid Balance Test" on **page 1122**.

VIEWING HYBRID BALANCE RESULTS

1. After running the Hybrid Balance Test, you can view the results in Message Print and manually change the setting if desired.

2. Maximize the Diagnostics Monitor window displaying the Expanded Message Print output. For the selected trunk, a free-form Message Print output displays the Echo Return Loss (ERL) for each of the 8 possible Hybrid Balance Settings and—based on the greatest ERL

value of the 8 settings—sets the Optimal Hybrid Balance Setting, as shown in the following example.


```
g Automated Hybrid Balance Test.
Balance Setting: 0   Raw Echo Value: 564   Effective ERL: 10.08 dB, Line ERL: 7.08 dB
Balance Setting: 1   Raw Echo Value: 380   Effective ERL: 13.51 dB, Line ERL: 10.51 dB
Balance Setting: 2   Raw Echo Value: 216   Effective ERL: 18.42 dB, Line ERL: 15.42 dB
Balance Setting: 3   Raw Echo Value: 300   Effective ERL: 15.56 dB, Line ERL: 12.56 dB
Balance Setting: 4   Raw Echo Value: 337   Effective ERL: 14.55 dB, Line ERL: 11.55 dB
Balance Setting: 5   Raw Echo Value: 488   Effective ERL: 11.34 dB, Line ERL: 8.34 dB
Balance Setting: 6   Raw Echo Value: 289   Effective ERL: 15.89 dB, Line ERL: 12.89 dB
Balance Setting: 7   Raw Echo Value: 756   Effective ERL: 7.54 dB, Line ERL: 4.54 dB
ed Hybrid Balance Test – Best Effective ERL: 18.42 dB  Optimal Hybrid Balance Setting: 2
 Receive Gain: 0 dB
g Automated Hybrid Balance Test.
Balance Setting: 0   Raw Echo Value: 819   Effective ERL: 6.84 dB, Line ERL: 6.84 dB
Balance Setting: 1   Raw Echo Value: 563   Effective ERL: 10.10 dB, Line ERL: 10.10 dB
Balance Setting: 2   Raw Echo Value: 305   Effective ERL: 15.42 dB, Line ERL: 15.42 dB
Balance Setting: 3   Raw Echo Value: 417   Effective ERL: 12.70 dB, Line ERL: 12.70 dB
Balance Setting: 4   Raw Echo Value: 476   Effective ERL: 11.55 dB, Line ERL: 11.55 dB
Balance Setting: 5   Raw Echo Value: 712   Effective ERL: 8.06 dB, Line ERL: 8.06 dB
Balance Setting: 6   Raw Echo Value: 428   Effective ERL: 12.48 dB, Line ERL: 12.48 dB
Balance Setting: 7   Raw Echo Value: 1088  Effective ERL: 4.37 dB, Line ERL: 4.37 dB
ed Hybrid Balance Test – Best Effective ERL: 15.42 dB  Optimal Hybrid Balance Setting: 2
Receive Gain: 0 dB
```

3. Record the pertinent results printed in the Message Print client.

4. *If you want to manually change the Hybrid Balance setting,* return to the DB Programming session. Select System – Devices and Feature Codes – Trunks – <***trunk extension tested***>. For the Hybrid Balance option, enter the value recommended for the Optimal Hybrid Balance Setting (see "Manually Changing the Hybrid Balance Setting" on **page 1125**).

5. Repeat this procedure and compare the greatest ERL values. Acceptable tolerance between the first and second runs is ±2 dB.

   • *If the second Hybrid Balance Test results are within tolerance,* the process is complete.

   • *If the greatest ERL in second run is more than the allowed tolerance,* run the test and set the Hybrid Balance value until you obtain consecutive ERL results within the ±2 dB tolerance.

NOTE	If the ERL (Echo Return Loss) measured on the trunk in question is out of specification, contact the service provider to resolve the issue. If the ERL is out of specification, the trunk could experience audio issues such as echo.

MANUALLY CHANGING THE HYBRID BALANCE SETTING

To view or change the Hybrid Balance line setting:

Select System – Devices and Feature Codes – Trunks – <***trunk number***>. The Hybrid Balance option shows the line setting, as shown in **Figure 67**. Select the current Value and scroll to the desired option to change the line setting.

Line Setting

ALARMS

Alarms are output as the result of continuous self-diagnostics run within the system and are a basic indicator that there is a problem or potential problem with the system. The severity and type of alarm determines the corrective action necessary to resolve the problem. A Major alarm indicates a problem that needs the immediate attention of a field service technician. Most Minor alarms an administrator can clear and do not require a service call. However, all alarms should be noted and monitored to determine if they occur on a regular basis, which may indicate a more severe problem. See the following section for definitions of alarm types.

In some instances, the corrective action for the condition requires contacting Mitel Technical Support personnel. The service technician is expected to be prepared and have the error information ready prior to calling Technical Support. For a complete list of the alarm messages and their corrective actions, refer to the *Message Print Diagnostics Manual,* part no. 550.8018.

This section contains the following information:

- **Alarm Types** below

- Network Alarms on **page 18-1128**

- Displaying Alarms on **page 18-1128**

- Alarm Queue on **page 18-1129**

- Clearing an Alarm on **page 18-1130**

- Responding to a Major Alarm on **page 18-1130**

ALARM TYPES

On the MiVoice Office 250 platform, alarms are grouped into the following categories:

- **Minor System alarms (000–019)**: These alarms indicate a Call Processing problem that normally are corrected without calling service personnel.

- **Minor Voice Processing alarms (020–039)**: These alarms indicate a voice processing problem that normally are corrected without calling service personal.

> **NOTE** Even when a voice processing alarm has been registered, the system may still function correctly.

- **Major System alarms (100–199)**: These alarms indicate a Call Processing problem that require attention from service personnel.

- **Major Voice Processing alarms (200–224)**: These alarms indicate a voice processing problem that require attention from service personnel.

- **Network alarms (225–244)**: These alarms indicate either Call Processing and voice pro-cessing problems that are generated from a remote node. These alarms are handled the same as the local alarm is handled. When a network alarm occurs, the local alarm (number) equivalent is displayed on the first line of the administrator's phone and the node where the alarm originated is indicated on the second line. What distinguishes a network alarm from a local alarm is the node information that appears on the second line of the phone's display.

NOTE	The actual alarm numbers 225–244 are used internally by the system and are not displayed on the administrator's phone. Instead, the administrator's phone shows the equivalent local alarm number between 000 and 224. Nothing appears in the Message Print output of a remote node, only on the local node is the Network alarm displayed.

NETWORK ALARMS

To allow one administrator to monitor multiple nodes, the system provides both system alarms and network-wide alarms:

- **Network-Wide Alarms**: When an event occurs that generates a network-wide alarm, the alarm is broadcast to every node in the system.

- **System Alarms**: System alarms appear only on the node on which the alarm was generated.

The following two flags in DB Programming determine whether a node broadcasts and/or receives network-wide alarms:

- The **Send Network Alarms** flag determines whether a node broadcasts alarms that occur on that node to the rest of the network. See "System Flags" on **page 751**.

- The **Receive Network Alarms** flag determines whether the node receives and displays alarms sent by other nodes in the network. The default state is No. See "System Flags" on **page 751**.

To differentiate between network-wide and local alarms, network-wide alarms appear on administrator phones preceded by "NET ALARM" and local system alarms are preceded with "SYS ALARM."

On remote nodes, network-wide alarms indicate the name of the node on which the alarm occurred. The node name is obtained from the username in Database (DB) Programming, if one is entered. Otherwise, only the node number is displayed.

DISPLAYING ALARMS

Depending on the settings in Minor System, alarm messages can be programmed to appear on the display of all administrator phones or on the primary attendant's display only. This is enabled by setting the Broadcast Alarms To All Administrators flag to Yes (see "System Flags" on **page 751**). Regardless of programming, major System alarm messages appear on all affected phone displays.

Network-wide alarms override system alarms on an administrator's phone display and on the LCD panel.

The display on an administrator's phone and the LCD panel on the unit function alike when displaying network-wide and system alarms. That is, alarms are automatically shown when the display is idle, and the alarms which appear on the LCD Panel are the same as those shown on an administrator's phone.

Only one alarm message is displayed on the LCD panel at any one time. Call Processing controls the generation of alarms, so if more than one alarm is generated the alarm with the

higher priority is displayed or replaces an alarm that is of a lesser priority. Because alarms are queued, the next alarm based on priority, is displayed once the previous alarm is cleared.

| **NOTE** | If the LCD panel displays ERROR, this is **not** a System alarm. See the following paragraph for more information. |

When the LCD panel displays ERROR after you attempt to make a system change using the LCD application, the log file should be examined to determine the cause of the error. View the rch_app_8.log file by accessing the Log Files Web page. Refer to Administrative Web Portal (AWP) Help for more information. The level of logging may need to be adjusted and the error recreated for the problem to show in the log files. All LCD application-related log entries have the form *<date> <time>* rch_app[*<log level>*]: LCD APP: *<log message>*.

ALARM QUEUE

This feature prioritizes system and network alarms based on severity and allows system administrator to view and handle critical alarms before addressing minor alarms. The administrator can then clear the individual alarm, or clear all the alarms in the queue (up to 30). When clearing alarms individually, the alarms are displayed in order of severity. The Emergency Alarm (A011) is the only priority 1 alarm. Other prioritized alarms have a 2, 3, or 4 priority, based on the severity of the alarm.

Not all alarms are prioritized. The numbered priority scheme is limited to alarms that can cause a major or minor system reset. Those alarms in the Alarm Queue that have a numbered priority are displayed before the alarms that do not have a numbered priority. (Priority 1 alarms have the highest priority.) Alarms that do not have a numbered priority are prioritized in the queue by date and time. When alarms are generated:

- The highest priority alarm is placed in the front of the queue, regardless of when lower priority alarms are generated. For example, if A114 (priority 3) and A116 (priority 4) are currently in the queue, but A119 (priority 2) is generated, A119 is placed first in the queue.

- Alarms with the same priority level are placed in the queue based on the time the alarm was generated. For example, if A010 (priority 3) is generated at 10:30 AM, and A012 (priority 3) is generated at 10:32 AM, A012 is placed in the queue after A010.

- If the queue contains 30 alarms, the oldest, lowest priority alarm is overwritten with the new alarm. For example, if the queue currently holds 30 alarms, 20 of which are priority 4, and a priority 3 alarm is generated, the oldest priority 4 alarm is overwritten.

- Repetitive alarms such as A125, are placed in the queue only once. If the alarm is regenerated, the alarm that is currently in the queue is overwritten with the new alarm data (if applicable) and time. For example, A125 is overwritten each time it is regenerated, which is every five minutes. This prevents the queue from being filled with duplicate alarms.

Table 172 shows which alarms have a numbered priority.

Table 172:   Alarms and Priorities

ALARM #	PRIORITY	ALARM #	PRIORITY	ALARM #	PRIORITY	ALARM #	PRIORITY
A010	Low	A026	Low	A112	Med	A128	Low

Table 172:   Alarms and Priorities

ALARM #	PRIORITY	ALARM #	PRIORITY	ALARM #	PRIORITY	ALARM #	PRIORITY
A011	Critical	A031	Low	A114	Med	A134	High
A012	Low	A032	Low	A115	Med	A135	High
A013	Low	A100	Low	A116	Low	A137	High
A014	Low	A101	Low	A117	Low	A138	High
A015	Low	A102	Low	A118	Low	A200	Low
A016	Low	A103	Low	A119	High	A201	Low
A017	Low	A104	Med	A120	Med	A202	Low
A018	Low	A105	Low	A121	High	A203	High
A020	Low	A106	Low	A122	High	A204	High
A021	Low	A107	Low	A123	High		
A022	Low	A108	Low	A124	Med		
A023	Low	A109	Med	A125	High		
A024	Low	A110	Med	A126	Med		
A025	Low	A111	Med	A127	Med		

CLEARING AN ALARM

An administrator can clear a network-wide alarm on the local node only or on every node in the network using their designated phone.

- **Clear Network Alarm (9851)**: Entering this feature code clears network-wide alarms on every node in the network, but does not affect system alarms. The Clear Network Alarm feature code may be entered on any node in the network, but the Send Network Alarms flag must be set for the administrator to clear alarms on other nodes in the network.

- **Clear System Alarm (9850)**: Entering this feature code clears all local and network-wide system alarm displays on your node.

NOTE	The LCD panel only displays alarms and cannot be used to clear an alarm. Only when the LCD application receives a message from Call Processing indicating the message is removed, is the LCD panel cleared.

RESPONDING TO A MAJOR ALARM

Since major alarms indicate that all or part of the system is inoperative, they require immediate attention from service personnel. If a system-wide failure is detected, a MAJOR ALARM message appears on all phone displays. If the major alarm message appears only on a group of phones, their associated module may have failed. If the major alarm message appears on

a single phone, the phone or its cabling may be defective. When a system-wide major alarm occurs, do the following:

1. DO NOT ATTEMPT TO REBOOT THE SYSTEM. Open a Web session and check if all applications are running. Typically, if an application is stopped, it is restarted by Call Processing, which can be observed by viewing the LCD panel messages. Check Message Print and save the output to a log file, if necessary.

2. If a Web session cannot be opened, check the network connection and ping the unit. Connect to the USB-B port and run online Monitor to view diagnostics.

3. Only as a last option should you reboot the system. Do not pull AC power. Reboot the system through the LCD Panel menu. If the system still does not recover from the alarm, use the troubleshooting charts beginning on "Troubleshooting CO Trunks" on **page 225** to try and identify the problem. If it is determined that the Processor Module or any other part is faulty, return it for repair and include any indicated error messages in the problem description.

| NOTE | When returning a faulty part, indicate all applicable error messages on the Material Return Authorization (MRA) tag. |

DIAGNOSTICS THROUGH DB PROGRAMMING

This section describes the diagnostic utilities available from the Database (DB) Programming interface:

- **Automatic Diagnostics Delivery** below

- **System Device Information** below

- Save/Restore Node Data on **page 18-1136**

AUTOMATIC DIAGNOSTICS DELIVERY

The Automatic Diagnostics Delivery (ADD) feature is a diagnostics utility available. When ADD is configured, the system automatically collects and sends diagnostic information to Mitel when it is triggered by the detection of an error that occurs within a set of defined parameters. Because the information is sent directly to Mitel Technical Support is able to retrieve system freezes without involving a technician, providing quicker response times.

Call Processing uses port 443 for communicating with the Mitel ADD server. If this port is blocked when attempting to send freezes, an error message is displayed in Message Print. ADD is available only in online Monitor (OLM) mode. To enable ADD, you must contact Mitel Technical Support.

SYSTEM DEVICE INFORMATION

There is an option on the View menu in Mitel DB Programming (View – **System Device Info**) that opens the System Device Info dialog box. This dialog box allows you to display all of the system devices in a centralized location for the system. The TDM and IP devices are displayed at the top of the dialog box, followed by the system's non-physical devices.

Note: 52xx/53xx IP phones awaiting PIN activation do not appear in the IP Device Status dialog box. See "Using PIN Activation for 52xx/53xx Phones" on **page 363** for details about the PIN Activation feature.

ASSOCIATED DEVICES AND REFERENCES

In database programming for any specific phone, the Associated Devices and References feature allows you to see the devices associated with a phone, mailbox, or hunt group. This feature also allows you to query various groups in the database to locate the associated references to the extension.

To search for associated devices or references:

Right-click the phone, mailbox, or hunt group, and then select **Associated Devices and References**. A dialog box, similar the one shown below, appears.
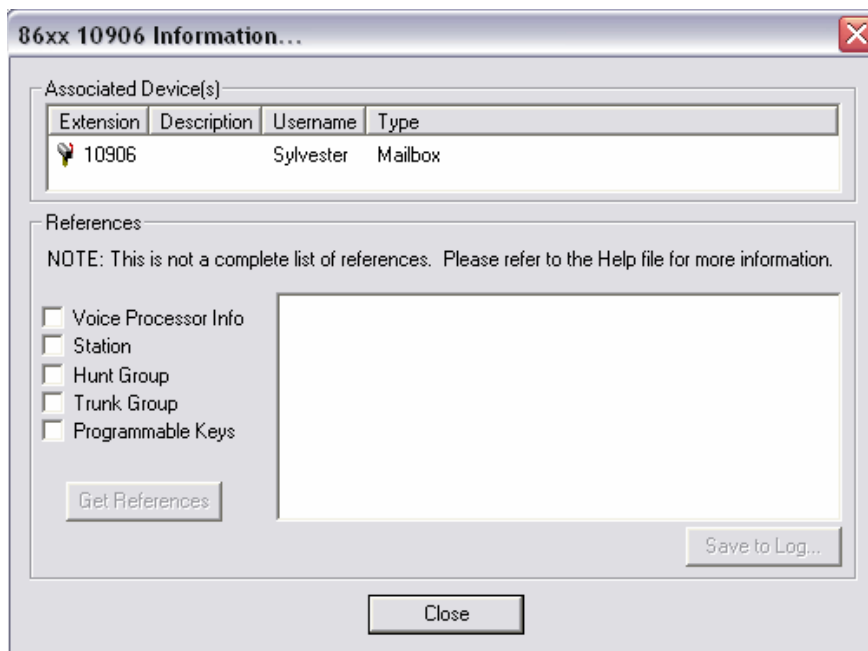

Table 173 shows the groups that you can query in the database for references, the type of information included in the groups, and the associated fields.

Table 173:   Database Query Options

GROUP	TYPE OF INFORMATION	FIELDS
Voice Processor Information	Applications AVDAP Information Call Routing Announcements Local Mailbox (Primary/Alternate Cascade Levels) Group List Members	Alternate Message Source Attendant Transfer Recall Destination Dial-0 Day Destination Type (Mailbox/Extension) Dial-0 Night Destination Type (Mailbox/Extension) System Administrator Mailbox Fax Delivery Destination Device (Extension)

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Table 173:   Database Query Options (continued)

GROUP	TYPE OF INFORMATION	FIELDS
Phone	Call Routing Entries Phones Emergency Day Outgoing Access List Emergency Night Outgoing Access List Forwarding Points	Ring-In (Extension) Attendant Message Center Transfer Recall Destination Device (Extension)
Hunt Group	Hunt Group Supervisors Hunt Group ACD Agent IDs Hunt Group Members Hunt Groups	Device (Extension) Announcement (Extension) Overflow (Extension) Recall (Extension)
Trunk Group	CO Trunk Group Day Answer Access List CO Trunk Group Day Multiple Ring In List CO Trunk Group Night Answer Access List CO Trunk Group Night Multiple Ring In List Trunk Group Day Outgoing Access List Trunk Group Night Outgoing Access List CO Trunk Groups	Device (Extension) Day Single Ring In (Extension) Night Single Ring In (Extension)
Programmable Keys	Programmable Keys Single Line Default Programmable Keys Keyset Default Programmable Keys Keyset KeyMap Keys	Device (Extension)

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PERIODIC DIAGNOSTICS

Periodic Diagnostics is a feature that extends and improves the functionality and checks and reconciliations performed on various resources in the system. Periodic Diagnostics is responsible for two areas of the system: system resources and the static database.

Periodic Diagnostics checks, reconciles, and/or repairs the following system elements:

- DTMF receivers

- Caller ID receivers and transmitters

- Speakerphone resources

- Connection IDs

- Voice channels

- IP resources

- System hardware

- Static database

- Off-node device information

- Unassociated mailbox information

- Temporary extensions

SYSTEM SOFTWARE PERFORMANCE STATISTICS

The Software Performance Statistics log contains software availability information. This information includes detailed performance and reset statistics.

By default, the system is configured to log software performance statistics. The information included in the log is used by Mitel Technical Support. The Internal Statistics Logging flag under System\Flags in OLM mode enables/disables this function. The default setting for this flag is **Yes**.

SAVE/RESTORE NODE DATA

The following sections provide database operations information.

ERROR INFORMATION

To capture related data when a system error occurs, open the Operations menu and select **Error Information...**. The following screen appears.
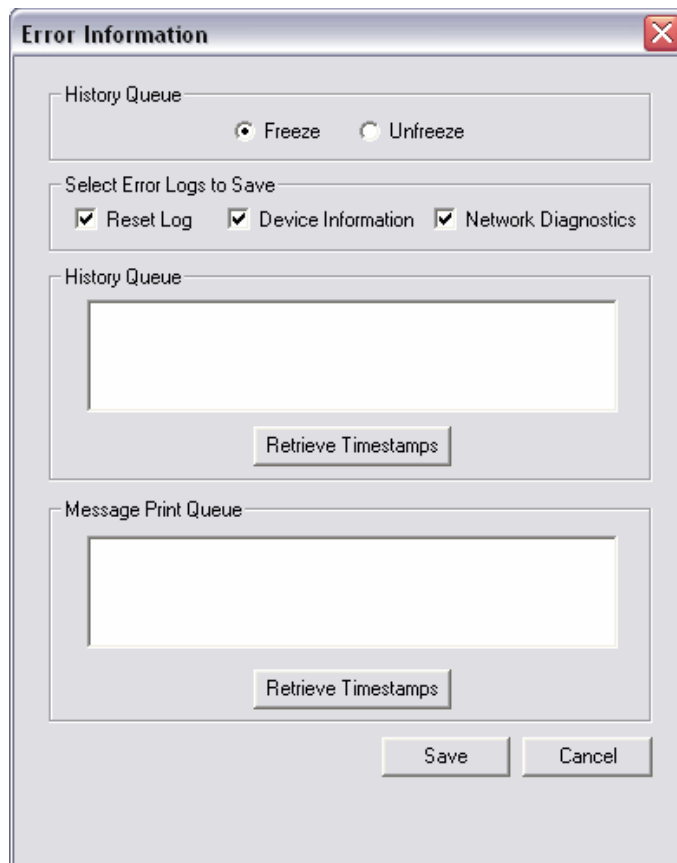
Select **Freeze** in the History Queue and perform the following steps:

1. Select the Error Log(s) to save in the freeze.

- **Reset Log**: What is included in this information?

- **Device Information**: What is included in this information?

- **Network Diagnostics**: When you select Network Diagnostics option before you complete a system freeze, the freeze includes a Network Diagnostics Log file with an extension of `.ndl`. Currently, this log file captures CP, IP Resources and IP settings, as well as insufficient bandwidth alarms. This information is included as one of several diagnostics dumps

2. Select History Queue and/or Message Print Queue and click on **Retrieve Timestamps** for the queues that you want to save.

3. Click **Save**. This prompts you for a location to save the information. When a freeze is performed the system automatically creates a variety of error logs and stores them in the location selected. By default, the number of freezes that can be saved is 30.

4. Once the freeze is completed you are prompted to unfreeze the database, select **Yes**.

IP DEVICE STATUS

The IP Device Status folder resides under the Operations menu at the Mitel DB Programming window. The model column describes the model of phone being displayed. From the Mitel DB Programming menu bar, click Operations and then **Get Status** to retrieve IP Device status.

When a phone comes online, it searches for a valid license. The license category column displays which category of license the phone has consumed. For more information, see "IP Device Status" on .

VOICE PROCESSING DIAGNOSTICS

The voice processing diagnostics that you use depends on what type of voice mail is running on the system. Refer to the *Voice Processing Diagnostics Manual*, part no. 550.8019.

With Unified Voice Messaging, the avdapmon log file can be reviewed through the Web page and if necessary, a freeze of the information can be taken using database programming. See "Error Information" on **page 1136**.

Running `avdapmon.exe` is useful when more information is needed to be displayed than what gets logged into the log file. To run AvdapMon from a computer, go to the **Start** menu, then select **Run**. Enter the file pathname, IP address of the system you are monitoring, and 4444 (port). The example below uses the default pathname.

`C:\InterTel\CS5000\AvdapMon\avdapmon.exe 172.xxx.xxx.xxx 4444` (v3.0 or earlier)

`C:\Program Files\Mitel\5000\AvdapMon\avdapmon.exe 172.xxx.xxx.xxx 4444` (v3.1 or later)

In Mitel DB Programming under Voice Processor/Maintenance, the Expanded Diagnostics flag has been enabled (Yes), to allow the user to enter commands outlined in the diagnostics manual. When the flag is enabled, the Voice Processor logs all diagnostics output (including alarms) generated by voice processing to a file on the computer. Using that file, you can troubleshoot problems dealing with message lamps, delayed messages, remote messaging, and so forth. Once the diagnostics flag is enabled, it can be left on at all times without affecting performance of the system.

Once connected, the AvdapMon window opens and you are prompted to enter a password. The password is blank and cannot be changed, press **Enter**. With the Expanded Diagnostics flag enabled, perform the desired command referred to in the *Voice Processing Diagnostics Manual*.

OTHER DIAGNOSTIC FEATURES

There are other diagnostics available in addition those mentioned in the previous sections that cannot be easily categorized or are a combination of diagnostic features. These include such things as diagnostic feature codes, audio diagnostics, and OLM command line.

This section contains the following information:

- **Administrator Phone Support** below

- **Diagnostics Feature Codes** below

- On-Line Monitor Command Line on **page 18-1142**

- LCD Panel Diagnostic Options on **page 18-1144**

- Resource Manager CPH Diagnostics Flag on **page 18-1144**

- Call Processing History (CPH) Freeze File Compression on **page 18-1145**

- History Queues/Log Files – Clearing on **page 18-1145**

- Traceroute on **page 18-1145**

ADMINISTRATOR PHONE SUPPORT

For troubleshooting purposes, diagnostics information is available from an administrator's phone. You can enable and disable diagnostics functions and also dump extension or node information to Message Print from an administrator's phone.

> **NOTE** Mitel recommends that you contact Technical Support for assistance with diagnostics issues.

DIAGNOSTICS FEATURE CODES

The Diagnostics Mode feature code (9900) [9100 in Europe] must be entered at the administrator phone to enable system diagnostic mode before the feature codes summarized in **Table 174** can be used.

Table 174:   Diagnostics Feature Codes

FEATURE NAME	CODE U.S. (EUROPE)	DEFINITION
Compression On/Off	9982 (9182)	Compresses call processing messages sent to DB Programming, speeding up transfers. You should not disable this feature unless instructed to do so by Mitel personnel.
Compression Statistics	9981 (9181)	Dumps various statistics related to the DB Programming compression algorithm. This feature should be used only when directed to do so by Mitel personnel.

Table 174:   Diagnostics Feature Codes (continued)

FEATURE NAME	CODE U.S. (EUROPE)	DEFINITION
Diagnostic – ASAI Snoop Off	9926 (9126)	Turns off the ASAI output to Message Print. ASAI is the protocol the system uses to talk to DB Programming and the AVDAP. Turning this feature on helps the Mitel Engineers debug the messaging between Call Processing and DB Programming or Call Processing and the AVDAP.
Diagnostic – ASAI Snoop On	9927 (9127)	Turns on the ASAI output to Message Print. ASAI is the protocol the system uses to talk to DB Programming and the voice processing system. Turning this feature on helps the Mitel Engineers debug the messaging between Call Processing and DB Programming or Call Processing and the AVDAP.
Diagnostic – Dump Extension	9933 (9133)	Allows the field technician to dump a device or structure when debugging a problem. The system dumps the IP Device Resource Manager information (the internal data structures for that extension) to Message Print. If you press the pound button (#) you are prompted for a command. The command allows you to enter an alphanumeric string of the structure to dump. See **page 1119** for details about IP Device Resource Manager.
Diagnostic – Dump Node Information	9936 (9136)	Pressing the Dump Node Information feature code dumps specified node information to Message Print for diagnostic purpose. When the system prompts for a node number, enter the applicable node number or zero (0) for all nodes within a network. If the node does not exist, an error message saying INVALID NODE NUMBER appears and prompts you for a node number again.
		After entering a node number, the system displays a confirmation message on the phone display. This helps you analyze which nodes are up and which nodes are down (the word 'down' does not mean that the node is completely down, it simply means the node is unreachable). For Example:
		If all nodes are up, the display shows ALL NODES ARE UP X. The 'X' represents the total number of the nodes.
		If node 3 and 5 of a 5-node network are down, the display shows # NODES DOWN 2 3 5.
Diagnostic – Heap Dump	9943 (9143)	This is used by Mitel software developers and cannot be used in Beta or Production software.
Diagnostic – Heap Statistics	9947 (9147)	This feature code outputs miscellaneous heap information to Message Print as well as putting up a message on the phone that indicates the percentage of available dynamic heap memory. This feature code is useful in determining if the system is losing heap memory and how quickly the system may be losing it.

Table 174: Diagnostics Feature Codes (continued)

FEATURE NAME	CODE U.S. (EUROPE)	DEFINITION
Diagnostic – ISDN View	9948 (9148)	This feature code is toggles through the three different ISDN view output modes.
		Entering the feature code the first time puts the ISDN view feature into headers only mode. In this mode the system outputs all ISDN messages to Message Print in header format (that is, it does not contain any ISDN information elements).
		Entering the feature code the second time puts the ISDN view feature into full mode. In this mode the system outputs all ISDN message to Message Print in full format (that is, each ISDN information element).
		Entering the feature code one more time turns this feature off.
Diagnostic – Major Reset	9962 (9162)	This is used by Mitel software developers and cannot be used in Beta or Production software.
Diagnostic – Mark As Leaks	9945 (9145)	This is used by Mitel software developers and cannot be used in Beta or Production software.
Diagnostic - Mark As Quiescent	9946 (9146)	This is used by Mitel software developers and cannot be used in Beta or Production software.
Diagnostic – Minor Reset	9964 (9164)	This is used by Mitel software developers and cannot be used in Beta or Production software.
Diagnostic – Network Freeze Zone System Histories Diagnostic – Network Unfreeze Zone System Histories	9939 (9139) 9989 (9189)	The system fault history for any freeze zone in the network can be halted (frozen) or re-enabled using these feature codes when diagnostics mode is enabled. The fault history can then be extracted from each zone and used by service personnel when troubleshooting the system.
Diagnostic – Network Groups	9963 (9163)	Allows an administrator to verify that the Network Groups on the local node are programmed properly. When 9963 is entered at an administrator's phone, the telephone system initiates pings from each Mitel IP device on the local node and determines if the other devices respond to the ping. If a device does not respond to the ping or if a firewall is detected, the system issues a Message Print message.
Diagnostic – Print Auxdata	9972 (9172)	(Not Programmable) Sends a report to a designated printer or file that shows system reset history information to be used for troubleshooting purposes.
Diagnostic – Print Message Log	9975 (9175)	(Not Programmable) Sends a report to a designated printer or file that lists system error messages to be used for troubleshooting purposes.
Diagnostic – Print Network Log	9976 (9176)	This feature code prints the network log to Message Print. This feature is useful in determining the system of a networked system.
Diagnostic – Query Node Traffic	9978 (9178)	Using this feature code you can query the status of various devices on the system based on the status of the traffic flags in DB Programming. The output is sent to Message Print.

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Table 174:   Diagnostics Feature Codes (continued)

FEATURE NAME	CODE U.S. (EUROPE)	DEFINITION
Diagnostic – Show Version	9928 (9128)	To check the call processing software version at an administrator's phone, you can enable diagnostics mode, then enter this feature code to view the version and date of the call processing software. However, feature code 9928 displays the firmware version of the phone if it is in SIP mode.
Diagnostic – SIP View	9987 (9187)	Allows the user to change the system wide SIP output value. Options include No Output (OFF), Headers (HEAD), and Full Output (FULL).
Diagnostic – Spare 1–3	9910–9912 (9110–9112)	This is used by Mitel software developers and cannot be used in Beta or Production software.
Diagnostic – System History	9974 (9174)	This is used by Mitel software developers and cannot be used in Beta or Production software.
Diagnostic – View Displays	9983 (9183)	This is used by Mitel software developers and cannot be used in Beta or Production software.
Program Database	9932 (9132)	Can be used for programming phone, system, and trunk parameters.
Seize Device	9973 (9173)	Used during troubleshooting to seize a specific trunk or phone by indicating the board number, port number, and device number.
System History – Freeze	9993 (9193)	The system fault history can be frozen or unfrozen using these feature codes when diagnostics mode is enabled. Fault history is used by service personnel when troubleshooting the system.
System History – Unfreeze	9998 (9198)	

ON-LINE MONITOR COMMAND LINE

The On-Line Monitor (OLM) provides a diagnostics view of the operating system for troubleshooting purposes. The OLM system diagnostic commands described in this section provide additional details about the system and help troubleshoot errors.

Use only the commands provided in this chapter unless otherwise directed to by Mitel Technical Support personnel.

The OLM is accessible from either a remote location through an Secure Shell (SSH) interface using a third-party application or locally through the USB-B port on the front of the chassis. The USB-B port provides access to the system when the IP network is down. The OLM shell requires appropriate drivers to be loaded before the feature can be accessed.

Access to the OLM shell requires a username and password. The default login for the username is `it5k` and the default password is `itpassw`. The "OLM>" should appear on the screen indicating that it is the OLM shell. Mitel recommends that you change this password at the earliest convenience.

The OLM shell provides access to commands in the olm_bin directory. Only the commands in the olm_bin directory are available to the OLM shell. Using the help command with no arguments provides a list of the available OLM commands along with a brief description. From the OLM shell, this help is accessible by typing the command followed by "-h." A question mark can also be substituted for "-h" for built-in and system diagnostics commands. If the command entered fails indicating an invalid option error, the option is not supported.

There are two commands "Exit" and "." built-in to the shell which are not executed from the olm_bin directory. Using either of these commands exits out of the OLM shell.

SYSTEM DIAGNOSTICS COMMANDS

System diagnostic commands are executed from the olm_bin directory and provide diagnostic information on the system. Commands denoted with an "*" allow additional arguments to be passed to them.

cls — Clear the terminal screen

free* — Display the amount of free and used system memory (free)

help (or ?) — Display the available commands or help on a specific command

ifshow — Display the status of the currently active interfaces (ifconfig)

ls* — List directory contents (ls | more)

netstat* — Display information about networking subsystem (netstat | more)

ping* — Send echo request to network hosts (ping)

ps* — Report a process status (ps)

route* — Display the kernel routing tables (netstat -rn)

top* — Display the top CPU processes (top)

vmstat* — Display virtual memory statistics (vmstat)

APPLICATION DIAGNOSTICS COMMANDS

Application diagnostic commands are executed from the olm_bin directory and provide diagnostic information on the individual application. These commands can be run on any of the following applications: T1, IP Resource Application (IPRA), Loop Start, Single Line and RCH. The application diagnostic commands allow additional arguments to be passed to them as listed in the table.

applogctrl — Enable/Disable Application log directories, change log file directories, and clear log files. The log files are stored in RAMDisk(/var/log/intl/) or Flash (/usr/local/intl/logs/). The following arguments can be used in conjunction with the application command.

-n *<AppName>* Name of the application, select from the following: t1_app, ls_app, iprc_app, sl_app, rch_app

-s *<Bay Number>* Specify the bay number the application is running on.

-e Enables logging.

-d Disables logging.

-l *<dir>* Changes the log file directory.

-c Clears the log.

appstatus — Display the basic execution status of an application

-n <*AppName*> Name of the application, select from the following: t1_app, ls_app, iprc_app, sl_app, rch_app

-s <*Bay Number*> Specify the bay number the application is running on.

-e Provides extended information

The following application command is applicable only to the IP Resource application.

iprc_diag_stats — Display IP Resource statistics information - device connection information, send and receive session information and so forth.

-a <*Bay Number*> Bay Number for IP Resource is usually 7 and 9 for expansion module.

-V Displays IP Resource version information.

-T <*PortID*> Displays phone information for the particular port ID. If PortID is ALL it displays information of all the phones in the system.

-t <*PortID*> Displays detail phone information for the particular PortID.

-S <*SessionID*> Displays Send Session Information for the Session ID provided. ALL provides details for all the send sessions on the IP Resource

-s <*SessionID*> Displays Audio Session Information for the Session ID provided

-R <*SessionID*> Displays Rx Session Information for the Session ID provided. ALL provided information for all the Rx session on the IP Resource.

-r <*SessionID*> Displays In-Time packet graph for the Session ID provided.

The following application command is applicable only to the Modem application.

modem_diag_stats — Display Modem statistics information in terms of transmit and receive speeds, echo level, Signal Quality, Signal to Noise Ratio information and so forth.

-a <Bay Number> Bay Number for modems is usually 8.

LCD PANEL DIAGNOSTIC OPTIONS

The liquid crystal display (LCD) panel displays system status and command messages on two lines of 14 characters each. The LCD displays system status information to determine the operating condition of the system, and it displays menus and options for programming, changing, or resetting the system.

For more information about LCD functions, refer to the Installation chapter in the *MiVoice Office 250 Installation Manual* .

RESOURCE MANAGER CPH DIAGNOSTICS FLAG

The Log Resource Manager Output flag exists in the System\Maintenance\History Queue folder to enable or disable the resource manager Call Processing History (CPH) diagnostics feature.

CPH is a collection of inputs and outputs used to diagnose problems that occur on the system. This flag can only be accessed in online Monitor (OLM) mode. By default, this flag is set to No.

> **NOTICE**
>
> Do *not* use OLM mode unless you are instructed to do so by support personnel.

CALL PROCESSING HISTORY (CPH) FREEZE FILE COMPRESSION

When a system freeze is performed, a snapshot of the memory-mapped files is taken and copied into a freeze directory. The directory contains a new Freeze_00 file which is always the most current. The CPH Freeze File Compression feature zips the freeze files into a compressed file that includes the node number and the start and end freeze dates and times in the zip file name. For example, `n12_06-02-04_123010_06-02-04_freeze.zip` represents a node 12 system freeze that occurred on June 2nd, 2004, between 12:30 PM and 10 seconds and 4:00 PM and 23 seconds. 160023_freeze.zip.

The time stamp includes hours, minutes, and seconds to handle situations if the Call Processor (CP) on the PM-1 should happen to freeze multiple times in the same minute. The zipped freeze file resides in the C:\ICP\FREEZE directory on the CPS. When a freeze occurs, the compression utility appears in a console window. After one to two seconds, the console window will minimize automatically. High system use may increase the delay before the system minimizes the window. When a freeze occurs, the compression utility appears in a console window. After one to two seconds, the console window minimizes automatically. High system use may increase the delay before the system minimizes the window. When Automatic Diagnostics Delivery (ADD) is enabled, the system sends the single zipped file.

HISTORY QUEUES/LOG FILES – CLEARING

System software allows the history queue to be cleared to baseline a site for monitoring purposes. To clear a history queue and log files, move/delete the Memory Map Files (`.mmf`), but do not delete the static_.`mmf` file. To clear the history queues and log files, use the Clear Error Information option in Mitel DB Programming while in OLM mode. When this operation is performed, the system generates a new Message Print "M6099 Corrupted Queue was Cleared," which confirms when the queue or log file has been cleared.

TRACEROUTE

The Traceroute application tests the communication path to any IP address in the network. For example, a technician could perform a traceroute from the IPRA to an IP phone to confirm the IPRA can communicate with it. The technician could also run a traceroute to any IP device in the network (a PC, some public IP address, and so forth). The traceroute commands lists the IP addresses of each router that exist along the path from the local host to the given IP address of a destination host on the network.

The method for acquiring this information relies on ICMP (Internet Control Message Protocol) Echo and Time Exceeded messages which some routers are programmed to not send or ignore. The traceroute will list a "no reply" when it encounters a router that ignores the request. A traceroute command is available from the IPRA with OLM consoles.

To run the traceroute command: Select the "Debug Session" option from a telnet console. At the prompt, invoke the command by entering "tracert" followed by an IP address in numeric form at the prompt.

EXTERNAL DIAGNOSTIC RESOURCES

This section describes the general functionality of the following tools:

- **Administrative Web Portal** below

- System Manager on **page 18-1148**

ADMINISTRATIVE WEB PORTAL

NOTES	Administrative Web Portal Pages support Microsoft® Internet Explorer® (IE) versions 8 and 9, and Mozilla Firefox versions 10 and 11.

Administrative Web Portal (AWP) is MiVoice Office 250 Web interface that provides a comprehensive view of the controller to gather diagnostic information about applications that are running on the system. AWP is initiated from any computer that has Internet access, allowing you to view and analyze a system without having to be on-site.

To access AWP:

1. In the address bar of a Web browser, type the IP address of the system that you want to access.

2. After communication is established with the system, type the username and password. By default, it5k is the username (which cannot be changed), and "itpassw" is the password.

For more information about AWP, refer to AWP Help.

Troubleshooting Administrative Web Portal

Table 175 summarizes troubleshooting strategies recommended for resolving Administrative Web Portal discrepancies.

Table 175: Administrative Web Portal Troubleshooting Tips

Symptom	Possible Cause	Corrective Action
A 404 Not Found error comes up when the user tries to access the Web page.	The user tried to access a page that does not exist on the server.	Use the links provided by the menu to access available pages.
A "Cannot find server" error comes up when the user tries to access the Web page.	The user typed the IP address incorrectly.	Make sure the IP address is the address of the system and was not typed incorrectly. Also, make sure the system is running, plugged in, and the lighted server is running.
A 401 Unauthorized error comes up when the user tries to access the Web page.	The user typed the username or password incorrectly.	Make sure the user name and password is for the correct system and was not typed incorrectly.

Table 175:   Administrative Web Portal Troubleshooting Tips (continued)

Symptom	Possible Cause	Corrective Action
The T1M-2 application does not show up on the AWP page for the corresponding expansion bay	The module might not be online.	Verify that the module is properly seated in the desired expansion bay and that the **Online** LED is green.
While making several outgoing or incoming calls, the status of the channels still shows up as idle.	The channels are not actually in-use.	Verify that incoming and outgoing access is programmed correctly in DB Programming and that the CO trunk groups contain the desired list of trunks to test. Also verify, when testing calls over private networking, that the correct route group is programmed properly.
After busying out a port, some channels still show up as active when accessing the channel status page.	Channels already actively on a call while the user attempts to busy out the port will go into the Pending Busy state.	When the calls have completed, the channels will then go to the Busy state.

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SYSTEM MANAGER

System Manager is a server-based application that centralizes management functions for the system and various peripheral components. For the MiVoice Office 250, System Manager uses a Web interface. With appropriate licensing, System Manager allows you to view system information such as Message Print, IP Resources, and trunk diagnostics.

System Manager can be configured to perform a variety of functions, one of which is the ability to freeze log files. For more information about the system interface, refer to the *System Manager Installation and Maintenance Manual*, part no. 835.2743.

Raw Commands

System Manager users have the option to view IP resource oversubscription statistics for a MiVoice Office 250 agent. The various types of resource statistics are displayed when commands are entered on the Raw Commands Web page. The following commands apply to the MiVoice Office 250s:

- IP DRM Resource Diagnostics

- IP Resource Diagnostics

- VoIP DSP Manager

- PS/Base Server Socket Statistics