Table of Contents

Chapter 1: About This Manual

Overview ..... Section 1.1
Chapters ..... Section 1.2
Chapter / Section Layout ..... Section 1.3
Icons and conventions ..... Section 1.4

Chapter 2: Product Description

Overview ..... Section 2.1
Macro Plexus System ..... Section 2.2
Micro Plexus System ..... Section 2.3
Standard Peripheral Cards ..... Section 2.4
Optional Peripheral Cards ..... Section 2.5
Additional Equipment ..... Section 2.6

Chapter 3: Getting Started

Overview ..... Section 3.1
Install Plexus Administrator ..... Section 3.2
Upgrading ..... Section 3.3
Installation Overview ..... Section 3.4
Configuration Overview ..... Section 3.5

Chapter 4: Installation

Installation Considerations ..... Section 4.1
Mounting Procedure ..... Section 4.2
Inserting Peripheral Cards ..... Section 4.3
AC power connection ..... Section 4.4
System Grounding ..... Section 4.5
Trunk Connections ..... Section 4.6
Extension Connections ..... Section 4.7
Plexus Key Telephone Installation ..... Section 4.8
Analog Telephone/Device Installation ..... Section 4.9
Operator Console Installation ..... Section 4.10
External Paging (Pager) Interface ..... Section 4.11
Music-on-hold interface ..... Section 4.12
Data Communication Interface ..... Section 4.13
System Check-Out ..... Section 4.14

Chapter 5: Plexus Configuration Wizard

Introduction to Plexus Configuration Wizard..... ..... Section 5.1
About Plexus Configuration Wizard .... Section 5.2
Using the Configuration Wizard .... Section 5.3
Configuration Report .... Section 5.4

Chapter 6: Software Configuration

Introduction .... Section 6.1
Software configuration - with Plexus Configuration Wizard . Section 6.2
Software configuration -
without Plexus Configuration Wizard .... Section 6.3
Determine the structure of the system .... Section 6.4
Physically insert peripheral cards into the cabinet..... ..... Section 6.5
Launch Plexus Administrator .... Section 6.6
Start a new configuration .... Section 6.7
Using an existing configuration file .... Section 6.8
Complete the peripheral card layout .... Section 6.9
Configure Auto-Attendant Interface (AAI) card .... Section 6.10
Configure Integrated Voice Processor (IVP) card..... ..... Section 6.11
Create users .... Section 6.12
Create user groups .... Section 6.13
Assign users to user groups .... Section 6.14
Create trunks .... Section 6.15
Create trunk groups .... Section 6.16
Assign trunks to trunk groups .... Section 6.17
Configure AEI extension ports .... Section 6.18
Intentionally Left Blank .... Section 6.19
Configure ATI trunk ports .... Section 6.20
Configure system parameters .... Section 6.21
Configure users .... Section 6.22
Configure user groups .... Section 6.23
Configure trunks .... Section 6.24
Configure trunk groups .... Section 6.25
Saving a configuration .... Section 6.26
Establishing a link .... Section 6.27
Uploading the configuration file .... Section 6.28
Updating the system clock .... Section 6.29
Updating Plexus key telephone programming..... ..... Section 6.30
Updating system XOS .... Section 6.31

Chapter 7: Key Telephone Software Configuration

Overview ..... Section 7.1
Entering Programming mode ..... Section 7.2
Entering alphanumeric characters ..... Section 7.3
Programming menus ..... Section 7.4
Exiting Programming mode ..... Section 7.5
System Parameters ..... Section 7.6
System Time ..... Section 7.7
System Speed Dial ..... Section 7.8
User ..... Section 7.9
User Group ..... Section 7.10
Trunk ..... Section 7.11
Extension Ports ..... Section 7.12
Int. Voice Mail ..... Section 7.13
CTi IP Address ..... Section 7.14

Chapter 8: Feature Reference

Overview of Features ..... Section 8.1
Feature Code List ..... Section 8.1.1
Function Code List ..... Section 8.1.2
Plexus System Feature List ..... Section 8.1.3
Plexus System Features List ..... Section 8.1.3

Chapter 9: IVP Configuration

System Overview ..... Section 9.1
The Integrated Voice Processor (IVP) Peripheral Card ..... Section 9.2
The Automated Attendant ..... Section 9.3
Voice Mailboxes ..... Section 9.4
System Parameters ..... Section 9.5
Voice Mail System Manager ..... Section 9.6
Setting up Hold Queue Intercepts ..... Section 9.7
Multi-Language Option ..... Section 9.8

Chapter 10: T-1 / E-1 CAS

Overview ..... Section 10.1
Features and Advantages ..... Section 10.2
Hardware Description ..... Section 10.3
Installation ..... Section 10.4
Configuration ..... Section 10.5
Status Monitoring & Troubleshooting ..... Section 10.6
Glossary ..... Section 10.7

Chapter 11: ISDN PRI

Overview ..... Section 11.1
Features and Advantages ..... Section 11.2
Hardware Description ..... Section 11.3
Installation ..... Section 11.4
Configuration ..... Section 11.5
Status Monitoring ..... Section 11.6
Glossary ..... Section 11.7

Chapter 12: ISDN BRI

Overview ..... Section 12.1
Hardware Description ..... Section 12.2
Installation ..... Section 12.3
Configuration ..... Section 12.4
Troubleshooting ..... Section 12.5

Chapter 13: Direct Inward Dialing (DID)

Overview ..... Section 13.1
DID ..... Section 13.2
Mapped DID ..... Section 13.3
Installation ..... Section 13.4
DID Configuration ..... Section 13.5
Mapped DID Configuration ..... Section 13.6

Chapter 14: Least Cost Routing

Overview ..... Section 14.1
Description ..... Section 14.2
Configuration ..... Section 14.3
Administration ..... Section 14.4

Chapter 15: Automated Attendant Interface (AAI)

Introduction ..... Section 15.1
Configure Auto-Attendant IDs ..... Section 15.2

Chapter 16: CTI Interface

Overview ..... Section 16.1

Chapter 17: Remote System Management

About Remote System Management ..... Section 17.1
Requirements ..... Section 17.2
Plexus System setup ..... Section 17.3
Plexus Administrator setup ..... Section 17.4
Connecting to the Plexus system ..... Section 17.5
Uploading a configuration file ..... Section 17.6
Updating the system clock ..... Section 17.7
Updating Plexus Key Telephone programming ..... Section 17.8
Closing the link and disconnecting ..... Section 17.9

Chapter 18: Hotel & Motel Package

Introduction ..... Section 18.1
System Administrator Privileges ..... Section 18.2
DSS-60/DDSS-60 Operator Console ..... Section 18.3
Simple Voice Mailboxes. ..... Section 18.4
Features ..... Section 18.5

Chapter 19: Plexus Specification

Macro system ..... Section 19.1
Micro system ..... Section 19.2

Chapter 20: FCC Information

FCC notification and repair information ..... Section 20.1
Part 15 of FCC rules-interference information ..... Section 20.2

Chapter 21: Product Warranties

Overview ..... Section 21.1
Product Warranty ..... Section 21.2
Return Material Authorization (RMA) Service ..... Section 21.3
Out-of-Warranty Service ..... Section 21.4
Miscellaneous ..... Section 21.5

Chapter 22: Customer Service

Overview ..... Section 22.1
Sales Questions ..... Section 22.2
Orders ..... Section 22.3
Technical Support ..... Section 22.4

About This Manual

1

1.1 Overview

The **Plexus System Manual** is made up of several different chapters to be contained in the product manual binder. Each chapter contains sections which present product and feature information or instruction on a key area of the setup process.

1.2 Chapters

The chapters explain how to install, configure, and use the Plexus Hybrid-Key Digital Telephone System as follows.

- "**Product Description**" - This chapter describes the various models and components in the Plexus product line.
- "**Getting Started**" - This chapter provides an overview of the steps involved in the setup process.
- "**Installation**" - This chapter addresses the physical installation of the telephone system.
- "**Software Configuration**" and "**Key Telephone Software Configuration**" - These chapters address the alternative approaches to configuring the telephone system.
- **"Feature Reference"** - This chapter presents detailed information on each feature of the Plexus system.
- "**Integrated Voice Processor**" - This chapter presents detailed information on configuring the Voice Mail and Automated Attendant options.
- "**T1/E1 CAS**", "**ISDN PRI**", and "**ISDN BRI**" - These chapters provide information on the digital trunk capabilities of the Plexus system and the software configuration required to integrate these trunks.
- "**Direct Inward Dialing**" and "**Least Cost Routing**" - These chapters describe these features in detail to facilitate the proper implementation and utilization of these new features.
- "**Automated Attendant Interface**" - This chapter describes and explains the Automated Attendant Interface card.
- "**Hotel / Motel Package**" - This chapter provides detailed information on the abilities and uses of the Plexus system in a hotel or motel environment.
- " **Specifications**", "**FCC Information**", "**Product Warranties**", "**Customer Service**", and "**Remote System Management**" - These chapters complete the manual, offering various supplementary information.

1.3 Chapter / Section Layout

Each chapter is laid out in several sections. The sections represent different topics within the chapter. Often during a description of a feature or aspect of the Plexus system, you will be referred to a separate chapter or section for more information as opposed to a page number. This is done to allow for compatibility of different versions of the manual and to easily allow updates to the manual without having to replace the entire book.

1.3.1 Chapter / Section numbering

1.3.1.1 Chapter

The chapters are numbered in order from 1 to 22.

1.3.1.2 Sections / Sub-sections

Sections are numbered from 1 on, within each chapter. The section will use the numbering format AA.BB.CC.DD.

AA = chapter number

BB = section number

CC = sub-section number

DD = sub-sub-section number

For example, this chapter / section has been laid out in the following order:

1	chapter
1.1	chapter.section
1.2	chapter.section
1.3	chapter.section
1.3.1	chapter.section.sub-section
1.3.1.1	chapter.section.sub-section.sub-sub-section
1.3.1.2	chapter.section.sub-section.sub-sub-section
1.3.2	chapter.section.sub-section
1.4	chapter.section

All references will be made to chapters and sections instead of page numbers. If a reference is made to a sub-section or sub-sub-section, it should be easily found by finding the chapter, section, and then sub-section, in that order.

1.3.2 Page numbers

Each page has been given a number, but no references are made to the page number itself. The page numbers are included to help serve as a guide. The page numbers use the format "page *X*-*Y*". *X* is the chapter number and *Y* is the page number within the chapter. Page numbers restart at 1 within each chapter.

1.3.3 Page Numbering

Page numbering is included at the bottom of every page. Page numbering will include the page number, the section name, and the section number. This is done to easily find the desired place in the manual quickly and easily.

Page Footer:

Note
Only section names and numbers will appear at the bottom of the page, not sub-sections or below. To find a sub-section, find the section it falls under and locate the subsection using the numbering format shown above.

1.4 Icons and conventions

Several icons and conventions are used throughout the Plexus documentation to highlight important points and offer helpful hints and tips. The icons and conventions are as follows:

Note
A Note provides additional information on the current topic.

Tip
Tips provide helpful hints and always appear with a light bulb icon in the margin.

Example
Examples provide additional insight by describing an application of the current topic or concept. Examples always appear with an e.g. icon in the margin.

Warning
Warnings about the proper handling of the Plexus system and its components. These are used to alert you about factors that could cause damage to the system and/or void warranties.

Software menus, options, and tabs appear in **bold** type.

Buttons are indicated by bold, sans-serif type: **Button**.

Keys are indicated by bold uppercase letters: **ENTER**, **DELETE**.

Product Description

2

2.1 Overview

The Plexus product line is comprised of two hybrid-key digital telephone systems: Macro Plexus and Micro Plexus. The systems are designed to satisfy a wide range of communications needs from home offices to medium-sized business offices. Both digital systems incorporate state-of-the-art digital technology for control processing and utilize highly integrated Pulse Code Modulation/Time Division Multiplexing (PCM/TDM) devices for voice switching.

The unique, universal card-slot arrangement affords the highest level of expandability and flexibility. The Plexus digital systems are designed such that the Plexus key telephones are fully compatible between the two systems and standard single-line telephone devices are supported without the need for any special adapters.

2.2 Macro Plexus System

2.2.1 Physical Description

The Macro Plexus system is based upon a proprietary digital backplane and universal card-slot arrangement. The Macro Plexus system has a metal card cage with 11 card-slots. When fully configured, and utilizing the Inter-Unit Link Interface (ILI) card, the Macro Plexus has the capacity for 128 universal ports (CO lines or telephone devices). Optional peripheral cards include the Integrated Voice Processor (IVP), the Auto-Attendant Interface (AAI), the Inter-Unit Link Interface (ILI), additional Analog Extension Interface (AEI), Digital Extension Interface (DEI), Analog Trunk Interface (ATI) cards, Direct Inward Dial Trunk (DID) cards, and Digital Trunk Interface (T1, E1, ISDN) cards.

Note
The Macro Plexus port capacity is raised to 188 ports when using E1 peripheral cards and 176 ports when using T1 peripheral cards.

Table 2 - 1 Macro Plexus Cabinet Size

Cabinet	Height	Width	Depth
Macro Plexus System	305 mm	438 mm	248 mm

2.2.2 Power Supply

The Macro Plexus system houses a built-in automatic-ranging/automatic-switching power supply that is able to tolerate a wide range of AC power inputs. This is ideal for environments where the power feed is unstable and susceptible to fluctuations. Macro Plexus systems shipped in the U.S. come with a standard 110V plug. For use outside of the U.S. (e.g., Europe, Africa), the manufacturer recommends changing the plug to a 220V plug or installing a modular outlet capable of accommodating a 110V plug as voltage adapters are not suitable for continuous use.

In the event of an AC power failure, the Macro Plexus system contains a built-in Uninteruptable Power Supply (UPS) that instantly switches to battery backup, preventing costly downtime. System operation is not disrupted when the system switches over to battery backup. While AC power is applied, the system applies a trickle charge to the battery.

Note
Batteries are not included.

2.3 Micro Plexus System

2.3.1 Physical Description

The Micro Plexus system utilizes a single motherboard system with plug-in facilities for two expansion cards. The Micro Plexus system and power supply are enclosed in separate plastic cases. When fully configured, a single Micro Plexus system has capacity for 12 CO lines and 24 telephone devices. Optional peripheral cards include the Analog Combination Interface (ACI), the Integrated Voice Processor (IVP), the Auto-Attendant Interface (AAI), the Inter-Unit Link Interface (ILI), and additional Analog Extension Interface (AEI), Analog Trunk Interface (ATI) cards, Direct Inward Dial Trunk (DID) cards, and Digital Trunk Interface (ISDN) cards.

The motherboard incorporates 4 trunk ports and 8 hybrid-key extension ports while a single optional peripheral card, the Analog Combination Interface (ACI), adds an additional 4 trunk ports and 8 hybrid-key extension ports. The Micro Plexus system also is available with either the Auto-Attendant or the Integrated Voice Processor integrated onto the motherboard, thereby freeing up the two plug-in facilities for other peripheral cards.

Table 2 - 2 Micro Plexus Cabinet Size

Cabinet	Height	Width	Depth
Micro Plexus System	368 mm	292 mm	99 mm
Micro Plexus Power Supply	254 mm	150 mm	90 mm

2.3.2 Power Supply

The Micro Plexus system has a compact external power supply with an advanced design. The power supply does not provide automatic voltage selection. Rather, the input power of the power supply is factory-set to either 110V or 220V.

In the event of an AC power failure, the Micro Plexus system contains a built-in Uninteruptable Power Supply (UPS) that instantly switches to battery backup, preventing costly downtime. System operation is not disrupted when the system switches over to battery backup. While AC power is applied, the system recharges the batteries.

Note
Batteries are not included.

2.4 Standard Peripheral Cards

2.4.1 Digital Switch Processor (DXP)

A specific slot in each system is reserved for the Digital Switch Processor card. This card contains the digital switch matrix and the main Central Processing Unit (CPU). An array of Digital Signal Processors (DSPs) provides various resources such as tone detection and generation. The fully digital architecture ensures high quality digital voice and data transmission.

The serial port located on the DXP card is used to connect to a Windows-based PC. Configuration files, developed using the Plexus Administrator Windows application, are uploaded via a serial link between the DXP card and the PC. Plexus Administrator can be used to configure all aspects of the system. Since non-volatile memory is used to store the configuration information, the system does not have to be reconfigured following a power shutdown.

The Switch Operating System (XOS) or firmware is stored in flash-memory devices on the DXP card to facilitate easy operating system upgrades. The upgrades are uploaded to the DXP card using Plexus Administrator in a manner similar to the configuration upload discussed above. For more information, refer to “Updating system XOS” - section 6.31.

2.4.2 Analog Trunk Interface Card (ATI)

The Analog Trunk Interface card provides either 4 or 8 analog trunk ports. The ports are designed to accept loop start lines from the CO and are fully compatible with Centrex Services and Caller ID (name and number) information. Other call enhancing facilities include disconnect supervision to CO signals like Clear Forwarding and Polarity Reversal.

2.4.3 Analog Extension Interface (AEI)

The Analog Extension Interface card provides 8 hybrid-key extension ports. *Hybrid-key* refers to the system’s ability to accept the input of Plexus Analog Key Telephones, ordinary phones, or single-line telephone devices at each analog extension port. The interface automatically adjusts for tone or pulse (rotary) dialing when an ordinary phone or single-line telephone device is connected. The single-line capability of the extension port accommodates fax devices, answering machines, modems, cordless telephones, door phones, and external voice mail systems. Unlike other systems, no specialized adapters are needed to accept such devices.

2.4.4 Digital Extension Interface (DEI)

The Digital Extension Interface card provides 8 digital extension ports. These ports are designed to operate with the Plexus Digital Key Telephones. Due to the digital signaling, these ports are not compatible with Plexus Analog Key Telephones or other analog phone devices (see Analog Extension Interface).

2.5 Optional Peripheral Cards

2.5.1 Analog Combination Interface (ACI) - Micro Plexus only

The Analog Combination Interface card is provided as an expansion option for the Micro Plexus system only. It contains 4 trunk ports and 8 hybrid-key analog extension ports and is inserted into one of the plug-in facilities.

The trunk ports are designed to accept loop start lines from the CO and feature Caller ID (name and number) detection and disconnect supervision.

Hybrid-key refers to the system's ability to accept the input of Plexus key telephones, ordinary phones, or single-line telephone devices at each analog extension port. The interface automatically adjusts for tone or pulse (rotary) dialing when an ordinary phone or single-line telephone device is connected. The single-line capability of the extension port accommodates fax devices, answering machines, modems, cordless telephones, door phones, and external voice mail systems. Unlike other systems, no specialized adapters are needed to accept such devices.

2.5.2 Inter-Unit Link Interface (ILI)

The Inter-Unit Link Interface card, when installed in each system, allows for the integration of any two systems, providing a seamless expanded switch. Connecting two Macro Plexus systems provides capacity for a total of 128 ports.

Each cabinet in an expanded configuration has the ability to operate as a stand-alone system. However, use of the ILI card provides other signaling information, enabling the linked units to cooperate as a single system.

2.5.3 Auto-Attendant Interface (AAI)

The Auto-Attendant Interface card provides an integrated four-channel auto-attendant supporting up to 4 twenty-second greetings. Each greeting may utilize up to 10 dial digit mappings (e.g. “Press 1 for Sales”).

2.5.4 Integrated Voice Processor (IVP)

The Integrated Voice Processor card provides integrated voicemail capabilities to the Plexus systems. Available models support from 4 to 24 channels. The base configuration of the IVP offers over 40 hours of recording time, 511 mailboxes, and the ability to utilize up to 10 simultaneous auto-attendant trees. The serial port on the IVP card is used to archive stored voice mail messages on the IVP card hard drive.

2.5.5 Digital Switch Processor with Computer Telephony Integration (DXP-CTi)

See Chapter 16 "CTi Interface"

2.5.6 Direct Inward Dial Trunk Interface (DID)

Direct Inward Dialing (DID) is service offered by telephone companies in which the last digits (typically two to four) dialed by the caller are forwarded to the system on a DID trunk. DID allows the system to have several inbound dialed numbers that route directly to different users, user groups, auto attendants, fax machines, etc. DID also allows the system to have more telephone numbers than it has trunks because not all telephone numbers will be in use at the same time.

See Chapter 13 "DID Interface"

2.5.7 Digital Trunk Interfaces (T1, E1, ISDN)

2.5.7.1 T1 / E1 CAS

Channel Associated Signaling (CAS) is a generic name for a signaling method used on T-1 and E-1 carriers. T-1 uses Robbed Bit Signaling (RBS). This method utilizes all 24 channels for voice, data, and signaling, offering 24 voice channels for the Plexus system. E-1 provides 30 'clear' channels for voice. The two remaining channels are reserved for framing and signaling. Each voice channel represents a digital trunk into the Plexus system.

See Chapter 10 "T1 / E1 CAS"

2.5.7.2 ISDN PRI

ISDN PRI is an interface between a user and a digital telephony network capable of delivering many types of services. PRI can be implemented on either a T-1 or an E-1 carrier. In both cases, the full bandwidth of the bearer or B-channels is available for the transmission of voice. T-1 ISDN PRI offers 23 voice channels. E-1 SDN PRI offers 30 voice channels. Each voice channel represents a digital trunk into the Plexus system.

See Chapter 11 " ISDN PRI"

2.5.7.3 ISDN BRI

ISDN BRI is a communication protocol that utilizes existing local loop wiring (i.e., standard 2-wire, single pair) to carry a digital signal. This signal can be split into two separate voice or data channels. A four port ISDN BRI card offers up to eight voice channels. Each voice channel represents a digital trunk into the Plexus system.

See Chapter 12 "ISDN BRI"

2.6 Additional Equipment

2.6.1 PVT-30 and PVT-30D analog key telephones

The PVT-30 series is comprised of two *analog* key telephone models designed for use with the Plexus Hybrid-Key Digital Telephone System. Both models offer 52 interface keys including 20 programmable keys, speakerphone, message waiting display, and many other feature keys. The programmable keys can be set to monitor extensions or trunks or provide access to features, functions, or speed dial numbers. Several of the programmable keys on the PVT-30D can be set to access features by default.

The PVT-30 D offers a Liquid Crystal Display (LCD) capable of displaying system status messages and Caller ID information.

2.6.2 PDT-30D digital key telephone

The PDT-30D is a *digital* key telephone model designed for use with the Plexus Hybrid-Key Digital Telephone System. This phone requires only a two wire (single pair) connection to the Plexus system. The single pair wiring can help cut installation costs. The phone is not polarity sensitive, which prevents polarity reversal problems. The PDT-30D offers 52 interface keys including 20 programmable keys, speakerphone, message waiting display, and many other feature keys. The programmable keys can be set to monitor extensions or trunks or provide access to features, functions, or speed dial numbers. Several of the programmable keys on the PDT-30D can be set to access features by default.

The PDT-30D offers a Liquid Crystal Display (LCD) capable of displaying system status messages and Caller ID information.

2.6.3 PVT-22 and PVT-22D analog key telephones

The PVT-22 series is comprised of two *analog* key telephone models designed for use with the Plexus Hybrid-Key Digital Telephone System. Both models offer 34 interface keys including 10 programmable keys, speakerphone, message waiting display, and many other feature keys. The programmable keys can be set to monitor extensions or trunks or provide access to features, functions, or speed dial numbers. Several of the programmable keys on the PVT-22 series can be set to access features by default.

The PVT-22D offers a Liquid Crystal Display (LCD) capable of displaying system status messages and Caller ID information.

2.6.4 PDT-22D digital key telephone

The PDT-22D is a *digital* key telephone model designed for use with the Plexus Hybrid-Key Digital Telephone System. This phone requires only a two wire (single pair) connection to the Plexus system. The single pair wiring can help cut installation costs. The phone is not polarity sensitive, which prevents polarity reversal problems. The PDT-22D offers 34 interface keys including 10 programmable keys, speakerphone, message waiting display, and many other feature keys. The programmable keys can be set to monitor extensions or trunks or provide access to features, functions, or speed dial numbers. Several of the programmable keys on the PDT-22D can be set to access features by default.

The PDT-22D offers a Liquid Crystal Display (LCD) capable of displaying system status messages and Caller ID information.

2.6.5 DSS-60 Analog Operator Console

The DSS-60 Analog Operator Console provides 60 programmable keys for use in monitoring extensions or trunks, provide access to features or functions, or as filter keys for special applications (e.g., Hotel and Motel). The DSS-60 Analog Operator Console is for use with PVT-30 and PVT-22 series analog key telephones.

2.6.6 DDSS-60 Digital Operator Console

The DDSS-60 Digital Operator Console provides 60 programmable keys for use in monitoring extensions or trunks, provide access to features or functions, or as filter keys for special applications (e.g., Hotel and Motel). The DDSS-60 Digital Operator Console is for use with PDT-30D and PDT-22D digital key telephones.

2.6.7 Plexus Door Phone

The Plexus Door Phone is a hands-free intercom device. The Door Phone has a single illuminated button to activate a call and a speaker to perform the hands-free speakerphone operation.

3

Getting Started

3.1 Overview

To set up a Plexus system for operation, the following two procedures must be completed:

1. Installation
2. Configuration

3.2 Install Plexus Administrator

Plexus Administrator can run on any Windows-based PC that meets the following system requirements.

3.2.1 System requirements

- Windows 95, 98, or NT
- Pentium-class processor
- 32 MB RAM (The Plexus Administrator application typically utilizes less than 10 MB of RAM while running.)
- 10 MB of available hard-disk space
- CD Rom (3.5” floppy disks available)
- Available serial port. For uploading configuration and switch operating system files.
- Color Monitor

3.2.2 Installation using CD Rom

Install the Plexus Administrator Windows application and the required Borland Database Engine as follows:

1. Insert the CD into the CD Rom drive.
2. The CD will run automatically. Follow the screen prompts.

3.2.3 Installation using 3.5" floppy disks

Install the Plexus Administrator Windows application and the required Borland Database Engine as follows:

1. Insert Borland Database Engine Disk 1 of 2 into the floppy drive.
2. Click on

.
3. Select **Run**.
4. Type **A:\Setup** or **B:\Setup** on the **Open** command line.
5. Follow screen prompts – Insert Disk 2 of 2 when prompted.

6 When complete, remove Disk 2 of 2 from the floppy drive.

7 Restart the computer.

8 Insert Plexus Administrator Disk 1 of 2 into the floppy drive.

9 Click on

..

10 Select **Run**.

11 Type **A:\Setup** or **B:\Setup** on the **Open:** command line.

12 Follow the screen prompts.

3.3 Upgrading

Before beginning with installation and configuration, it may be necessary to upgrade the Plexus Administrator Windows application. To assure compatibility and peak performance, it is always best to utilize the most current version. Determine the existing Plexus Administrator version according to the instructions included in this section and compare the version number with the version available on the Internet (www.bbstelecom.com).

3.3.1 XOS Upgrade Procedure

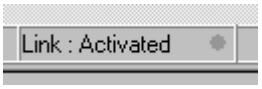
It may be necessary to upgrade the Switch Operating System (XOS) to an XOS compatible with your System Administrator software or to a newer version of the XOS for system performance.

3.3.1.1 Determining the Version of the Existing XOS

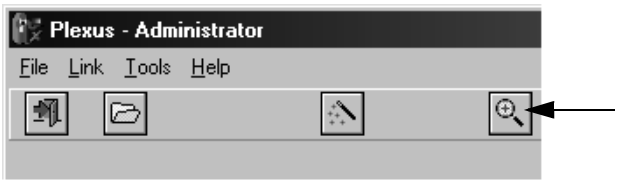
To determine the version of the existing XOS, proceed as follows:

1. Launch the Plexus Administrator version 2.0 Windows application.
2. Set up the link (see "Setting Up a Link" - section 3.3.3.1).
3. From the **Link** menu, select **Open**.
4. You will be prompted to enter a password. This will be the System Administrator Password for the system to which you are linking. If this is a new system, leave the Password as the default 123456.

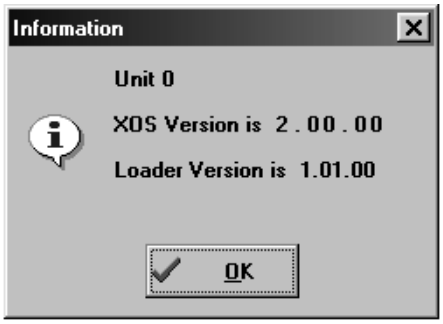
5. Enter the Password and click **OK**.
6. The link indicator at the bottom of the screen should indicate **Link: Opened** and the LED image should appear green.



7 Click on the magnifying glass icon.



8 If the current XOS version is a version 2.0 XOS, the XOS version will appear in a pop-up window.



9 If the current XOS version is a 1.02 XOS, the pop-up window will state "This version of Plexus Administrator is not compatible with the current version of XOS". You must upgrade the XOS using Plexus Administrator 1.02 (see“Upgrading version 1.02 XOS with a version of 2.0 XOS” - section 3.3.2)

3.3.2 Upgrading version 1.02 XOS with a version of 2.0 XOS

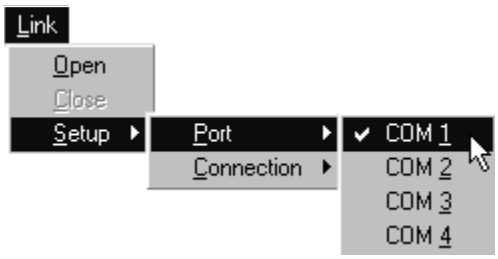
Note
For existing systems: Before upgrading XOS, ensure you have the system configuration saved on your computer. You will need to reload the configuration once the XOS upgrade is complete. If upgrading the XOS in conjunction with using a new version of Plexus Administrator software, see “Upgrade configuration using Dbconv.exe” - section 3.3.5.

3.3.2.1 Setting Up a Link

1 Connect the serial port on the DXP card (labeled “RS-232”) to an available COM port on a PC using a 9-pin serial cable (not included).

2 Launch the Plexus Administrator **version 1.02** Windows application. For instructions, see “Launch Plexus Administrator” - section 6.6.

3 Choose the COM port to which the serial cable is connected, by selecting **Setup** and **Port** from the **Link** menu (as shown below).



Note
The default Connection setting, Direct (**Connection**, **Direct**), applies to a serial connection. Modem connections apply only when performing Remote System Management.

Tip
The chosen COM port must be an *available* COM port. Read below to determine the availability of a COM port.

3.3.2.2 Serial and COM ports

Most computers provide two serial ports for use by serial devices. Serial ports are either 9-pin or 25-pin external ports. Each serial port has an associated COM port. Microsoft Windows supports COM 1, COM 2, COM 3, and COM 4. Generally, the two serial ports are pre-configured as COM 1 and COM 2. Internal devices such as modems may utilize a COM port. Should an internal device utilize COM 1 or COM 2, the associated serial port will be unavailable. Should an internal device utilize COM 3 or COM 4, the serial ports are available as long as the COM port utilized by the internal device has been assigned other than the default IRQ.

3.3.2.3 IRQs

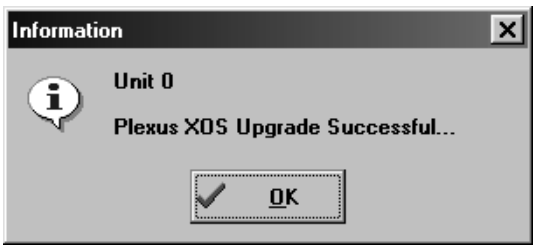
An interrupt request (IRQ) is a unique number (between 0-15) assigned to each device in a computer. The unique number assigned to a device enables the computer's processor to manage which device is receiving its attention. If devices share an IRQ, a conflict arises as the processor does not know which device to attend to. COM 1 and COM 3 share a default IRQ. COM 2 and COM 4 share a default IRQ. In order to simultaneously use COM 1 and COM 3 or COM 2 and COM 4, one of the COM ports must be assigned a different IRQ (i.e., other than the default).

A. 25-pin Serial Ports

Should the computer's only available serial port be a 25-pin port, a 9-25 pin adapter must be used. 9-pin serial cables and 9-25 pin adapters may be purchased at most electronics and computer supplies stores.

3.3.2.4 Uploading the XOS File

1. Launch the Plexus Administrator Windows application.
2. From the **Link** menu, select **Open**.
3. The link indicator at the bottom of the screen should indicate **Link: Opened** and the LED image should appear green
4. From the **File** menu, select **New**.
5. Choose the appropriate system type (e.g. Macro or Micro).
6. Right mouse click on the DXP card and select **XOS Upgrade**.
7. Locate the most recent.XOS file.
8. Click on **OK.**
9. It may take several minutes for the upload to complete.
10. When successfully completed, the system will notify you.



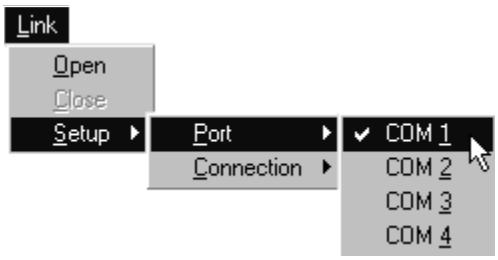
11. Click **OK**.
12. Shut down Plexus Administrator **version 1.02**. To upload the configuration or make any changes to this system, you will need to run Plexus Administrator version 2.0. See Chapter 6, "Software Configuration".

3.3.3 Upgrading version 2.0 XOS with a newer version of 2.0 XOS

Note
For existing systems: Before upgrading XOS, ensure you have the system configuration saved on your computer. You will need to reload the configuration once the XOS upgrade is complete

3.3.3.1 Setting Up a Link

1. Connect the serial port on the DXP card (labeled "RS-232") to an available COM port on a PC using a 9-pin serial cable (not included).
2. Launch the Plexus Administrator version 2.0 Windows application. For instructions, see "Launch Plexus Administrator" - section 6.6.
3. Choose the COM port to which the serial cable is connected, by selecting **Setup** and **Port** from the **Link** menu (as shown below).



Note
The default Connection setting, Direct (**Connection**: **Direct**), applies to a serial connection. Modem connections apply only when performing Remote System Management.

Tip
The chosen COM port must be an *available* COM port. Read below to determine the availability of a COM port.

3.3.3.2 Serial and COM ports

Most computers provide two serial ports for use by serial devices. Serial ports are either 9-pin or 25-pin external ports. Each serial port has an associated COM port. Microsoft Windows supports COM 1, COM 2, COM 3, and COM 4. Generally, the two serial ports are pre-configured as COM 1 and COM 2. Internal devices such as modems may utilize a COM port. Should an internal device utilize COM 1 or COM 2, the associated serial port will be unavailable. Should an internal device utilize COM 3 or COM 4, the serial ports are available as long as the COM port utilized by the internal device has been assigned other than the default IRQ.

3.3.3.3 IRQs

An interrupt request (IRQ) is a unique number (between 0-15) assigned to each device in a computer. The unique number assigned to a device enables the computer's processor to manage which device is receiving its attention. If devices share an IRQ, a conflict arises as the processor does not know which device to attend to. COM 1 and COM 3 share a default IRQ. COM 2 and COM 4 share a default IRQ. In order to simultaneously use COM 1 and COM 3 or COM 2 and COM 4, one of the COM ports must be assigned a different IRQ (i.e., other than the default).

A. 25-pin Serial Ports

Should the computer's only available serial port be a 25-pin port, a 9-25 pin adapter must be used. 9-pin serial cables and 9-25 pin adapters may be purchased at most electronics and computer supplies stores.

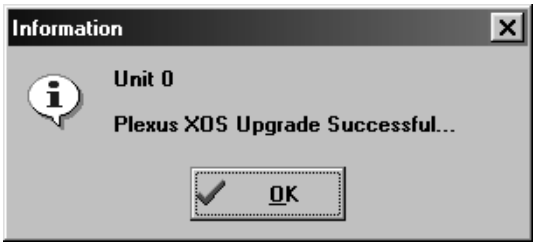
3.3.3.4 Uploading the XOS File

1. Launch the Plexus Administrator Windows application.
2. From the **Link** menu, select **Open**.
3. The link indicator at the bottom of the screen should indicate **Link: Opened** and the LED image should appear green
4. From the **File** menu, select **New**.
5. Choose the appropriate system type (e.g. Macro or Micro).
6. Right mouse click on the DXP card and select **XOS Upgrade**.

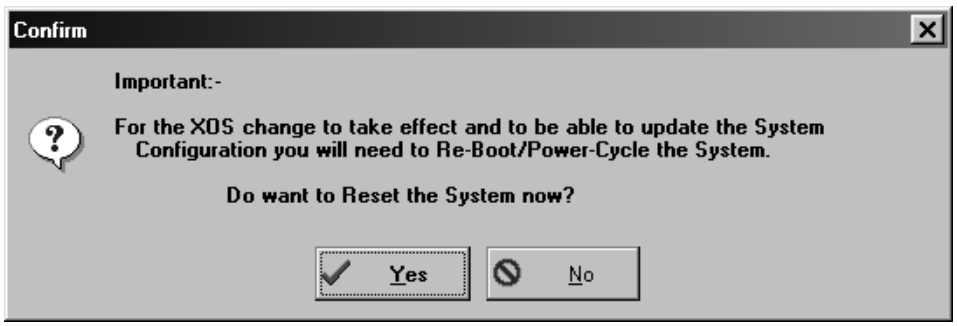
7 Locate the most recent.XOS file.

8 Click on **OK.**

9 It may take several minutes for the upload to complete. When complete, the system will notify you.



10 When successfully completed, the Plexus Administrator will prompt whether to re-boot the system now or wait. Select either.



Note
The XOS upgrade will not take effect until the Plexus system is re-booted or power cycled. This allows for an XOS upgrade to be entered but the actual resetting of the Plexus system to wait until the system is not in use.

11 If the Plexus was configured prior to the XOS upgrade, it will be necessary to reconfigure the Plexus and/or reload the configuration (.zdb) file once the Plexus system has been re-booted or Power cycled.

3.3.4 Plexus Administrator Upgrade Procedure

To upgrade Plexus Administrator, simply install the newer version to the default directory provided by the installation program. If the you want to use the default directory (e.g., C:\Plexus2.00), accept the default directory during the upgrade installation. Otherwise, determine the directory where Plexus Administrator should be installed. It is recommended you keep both the older and newer version of Plexus configuration software in case of system maintenance of older (not yet updated) Plexus systems.

Note
Configurations that were created with a previous version of Plexus Administrator may not be compatible the newer version.

Always backup the current installation of Plexus before installing a newer version.

3.3.4.1 Determining the Version of the Existing Plexus Administrator

To determine the version of the existing Plexus Administrator, refer to the installation disk label or proceed as follows:

1. Launch the Plexus Administrator Windows application.
2. Select **About** Plexus **Administrator** from the **Help** menu.

3.3.5 Upgrade configuration using Dbconv.exe

Configurations created with Plexus Administrator version 1.02 will not be compatible with Plexus Administrator version 2.0. To convert these configurations to a compatible format, a program is available called Dbconv.exe. To run Dbconv.exe, proceed as follows:

1. Open the folder used in “Install Plexus Administrator” - section 3.2 (default is C:\Plexus-2.00).
2. Find the program Dbconv.exe (on some systems, the program will appear as Dbconv)
3. Double click the icon to run the program.
4. Follow the instructions on the screen.

5 When finished, it may be desirable to modify the configuration to implement new features or functionality available in Plexus Administrator version 2.0. To modify the configuration, follow the instructions in “Using an existing configuration file” - section 6.8.

3.4 Installation Overview

Installation entails the physical setup of the various hardware components of the Plexus system, the wiring, and the connection of the telephone devices. The installation procedure includes:

- mounting the cabinet
- grounding the cabinet
- inserting peripheral cards
- wiring the CO lines to the trunk ports
- wiring the stations to the hybrid-key extension ports
- connecting ordinary telephones
- connecting Plexus Key Telephones
- connecting operator consoles
- connecting other single-line telephone devices (e.g., fax machine, modem, credit card scanner)

Note
Before inserting peripheral cards into the Plexus cabinet, refer to chapter 4 "Installation".

3.5 Configuration Overview

Configuration entails customizing the system with regard to users, user groups, trunks, trunk groups, etc. Configuration of the Plexus system is accomplished through the Plexus Configuration Wizard or through the Plexus Administration software. Chapter 5, "Plexus Configuration Wizard" and chapter 6, "Software Configuration" provide detailed instructions.

Errata

Section 3.2—Install Plexus Administrator (page3-2)

3.2.1 System requirements

- Windows 95, 98, NT.

Should read

- Windows 95, 98, 2000, NT, ME and XP(1)

- CD Rom (3.5" floppy disks available)

Should read

- CD ROM (3.5" floppy disks NOT available)

3.2.3 Installation using 3.5'' floppy disks

This section is NOT applicable.

Footnote:

1. **No *Help* file is available under Windows 2000, ME and XP systems.**

Installation

4

4.1 Installation Considerations

4.1.1 Dealer Tips

Thorough planning of an installation aids in a smooth transition of phone service and a satisfied customer. Dealers/Installers should consider the following when consulting with a customer prior to the installation of a Plexus Hybrid-Key Digital Telephone System:

- Select a suitable location for the system.
- Determine the number of trunk ports and extension ports required by the customer.
- Develop a floor plan.
- Determine the location and number of stations of each type, including key telephones, ordinary telephones, fax machines, modems, etc. Mark the floor plans accordingly. *Only one key telephone set is allowed per extension port. It is not possible to attach multiple key telephones to a single extension port.*
- Determine the location(s) of the operator console(s). Mark the floor plans accordingly. Note that Plexus Operator Consoles utilize an extension port.
- Collect detailed configuration information (e.g., User Names, User Groups, Trunk information, etc.) so that the system may be configured either before, or while, the system is being installed.
- Arrange for station cabling and power cabling (if necessary).
- Connect the system to surge protection to provide additional lightning protection (beyond that which is provided on the CO lines by the local telephone company). See “Lightning Protection” - section 4.5.2 and “Surge Protection Considerations” - section 4.6.2 for more information.

Note
Installers should be trained and thoroughly familiar with the basic components of the system before attempting installation of the product.

4.1.2 Site Planning

The most basic, yet most critical, consideration in the installation of the Plexus system is the selection of a suitable location to mount the cabinet. The following points should be considered when choosing a location:

- Sufficient space should be allowed for access to the cards within the cabinet and to the Main Distribution Frame (MDF) panel, punchdown block, etc.
- Location where the CO lines are terminated.
- The relative location of the majority of the telephone sets (stations); in order to minimize the length of cable runs between the stations and the Key Service Unit (KSU).
- The KSU, MDF panel, and punchdown block should be located in an electrically noise-free environment, isolated and shielded from equipment that generates Electro-Magnetic Interference (EMI) and Radio-Frequency Interference (RFI) radiation.
- When installed in locations prone to lightning strikes, adequate provisions should be made for lightning protection on the power line, CO lines, and any station cable runs outside the building.
- The install location should be well ventilated, having an optimum temperature range of 60° to 80° F and a relative humidity range of 5 to 90% (non-condensing).
- The install location should be well lit for installation and maintenance of the system.
- Any hazardous or flammable materials should be removed from the vicinity.
- The immediate area should not be subject to flooding or excessive moisture.

4.1.3 Tools and Hardware

Ensure that the following items are available for installation purposes:

- fasteners: wood screws (¼" x 1" round head), toggle bolts, or wall anchors
- screwdriver
- electric drill - if prepared holes are required
- connecting tool - for fastening wires to the connector block
- crimping tools - for the modular plugs
- volt/ohm meter
- 10 gauge wire - for grounding

4.1.4 Installation Notice

Be aware of the following precautions applicable to the installation of equipment that is to be directly connected to the telephone network:

- Do not perform telephone wiring during a lightning storm.
- Do not install telephone jacks in wet locations, unless the jack is specifically designed for wet locations.
- Do not touch uninsulated telephone wires or terminals, unless the telephone line has been disconnected at the network interface.
- Use caution when installing or modifying telephone lines.

4.2 Mounting Procedure

Plexus systems may be wall-mounted, rack-mounted, or placed on a tabletop.

4.2.1 Wall-Mounting Procedure

Both Macro and Micro systems may be wall-mounted. Wall-mounting templates are included for each cabinet.

1. If a backboard is required at the mounting location, attach it securely to provide a stable mounting surface for the Plexus system.
2. Refer to the mounting template (included in the product manual) to determine the placement of the 4 mounting screws.
3. Drill holes, corresponding to the size of the fastener (e.g., wood screws, toggle bolts), in the mounting surface. If necessary, prepare the holes with inserts, anchors, or other attachment devices.
4. Apply the 4 fasteners into the mounting surface and tighten them to within approximately 5/16” (8 mm) of the surface.
5. Place the cabinet on the fasteners using the mounting holes located at the rear of the cabinet. Note that the holes are elongated with an enlargement at one end. This feature allows the cabinet to snap down on the screws to secure the mounting.

4.2.2 Rack-Mounting Procedure

Macro systems may be rack-mounted on standard 19-inch racks. Hardware, including brackets and screws, is not included, but may be special-ordered Please call Customer Service for more information.

1. Securely attach the rack ears (brackets) to the Macro cabinet with the provided screws.
2. Place the cabinet into the rack. The screw holes in the brackets must line up with the screw holes on the face of the rack.
3. Securely fasten the brackets to the rack with the provided screws.

4.2.3 Tabletop Procedure

Both Macro and Micro systems may be placed on and operated from a tabletop. Make sure that the table utilized is stable and capable of bearing the full weight of the system.

4.3 Inserting Peripheral Cards

4.3.1 Installation Notices

- Key telephone configuration can only be undertaken after all standard and optional peripheral cards have been inserted into the cabinet.
- **Before installing peripheral cards, the Plexus system must be completely powered down.** It is required that the system be unplugged during card installation.
- The system cards are sensitive to static electricity and should be handled by the edges only. Never touch the components on a card.

Warning
Do not install or remove any peripheral card into a Plexus system while the system is on. Power off the Plexus system before doing any card installation or removal. Failure to do so can damage system components and will void warranties.

4.3.1.1 Allowable Slot Configuration

Some peripheral cards may only be inserted into certain card-slots. The allowable card-slot configuration for each system is indicated below:

Slot	Macro System
0	AEI,DEI, ATI, AAI, IVP*, DID, ISDN BRI
1	AEI,DEI, ATI, AAI, IVP*, DID, ISDN BRI
2	AEI,DEI, ATI, AAI, IVP*, DID, ISDN BRI
3	AEI,DEI, ATI, AAI, IVP*, DID, ISDN BRI
4	AEI,DEI, ATI, AAI, IVP*, DID, ISDN BRI
5	AEI,DEI, ATI, AAI, IVP*, DID, ISDN BRI
6	AEI,DEI, ATI, AAI, IVP*, DID, ISDN BRI
7	AEI,DEI, ATI, AAI, IVP*, DID, ISDN BRI
8	AEI,DEI, ATI, AAI, IVP, DID, ISDN BRI, T-1/E-1 CAS, ISDN PRI, ILI
9	DXP
10	IVP, T-1/E-1 CAS, ISDN PRI
Slot	**Micro System**
Front	ACI, AEI, DEI, ATI, DID, AAI, ISDN BRI, IVP
Rear	ACI, AEI, DEI, ATI, DID, AAI, ISDN BRI, IVP, ILI

* = Only 4 or 8 port IVP cards may be installed in slot 0-7.

Note
If a T-1 or E-1 is installed in slot 10, slot 8 may only contain another T-1 or E-1 or an Inter-Unit Link Interface (ILI) card. If the T-1 or E-1 is installed in slot 8, no similar limitations exist.

4.3.1.2 Inserting the Peripheral Cards

Insert peripheral cards as follows:

1. Carefully remove the peripheral card from the packaging.

Note
The cards are sensitive to static electricity and should be handled by the edges only. Never touch the components on a card.

2. Place the edges of the card inside the top and bottom card guides within the Plexus system cabinet.
3. Gently slide the card in until it completely seats into the backplane. There will likely be an increase in pressure and a slight noise as the card is seated into the backplane.

Warning
Do not install or remove any peripheral card into a Plexus system while the system is on. Power off the Plexus system before doing any card installation or removal. Failure to do so can damage system components and will void warranties.

4.4 AC power connection

4.4.1 Primary Power Supply

Employ a dedicated 117VAC 15 AMP circuit, with a third-wire ground, supplied to a standard electrical outlet (NEMA 5-15R) for the AC power connection.

Note
Refer to chapter 2, "Product Description" or chapter 19, "Specifications" for information on the Macro Plexus and Micro Plexus system power supplies.

For added equipment protection:

- Connect a plug-in power line surge protector between the Plexus power cord and the AC outlet.
- Thoroughly check out the installation before connecting the Plexus power cord to AC power (i.e., an AC outlet or a surge protector outlet).

4.4.2 Battery Backup / UPS (Uninteruptable Power Supply)

4.4.2.1 Description

Plexus Systems provide an internal Uninteruptable Power Supply (UPS). Batteries are not included and must be added to the Plexus for the UPS to work properly.

Note
It is strongly recommended that the Plexus systems internal UPS be connected to a 24-volt battery source in order to maintain system operation during a power failure.

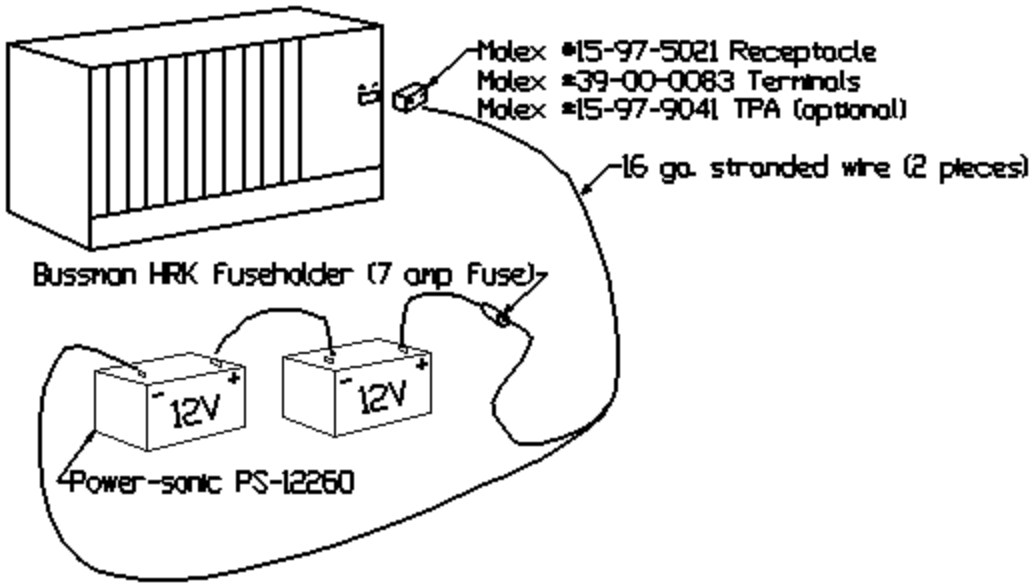
The power supply module provides a common equipment interface connector (labeled “24 Vdc”) for the connection of an optional external battery assembly. The required adapter cable is included with each Plexus system.

Before connecting an external battery assembly to the connector, be sure that the Plexus power cord is connected to an electrical outlet. This enables internal protection circuitry that may prevent damage resulting from an improper connection.

During AC operation, recharging current is provided to the common equipment interface to maintain the voltage potential of the external battery assembly. The recharging circuit may not provide an adequate charge if an installed battery assembly has a current rating of greater than 26 Ah.

An external battery assembly may require several hours to completely recharge after it has been completely discharged and, in some cases, when initially installed. Refer to the documentation supplied with the external battery assembly that is being used.

The recommended configuration consists of two 12-volt batteries connected in series as shown below.



The length of time that the system can operate on battery power is primarily determined by the characteristics of the particular battery being used. The battery draw of the Plexus systems is 6 A @ 24V max or 2.5 A @ 24V nominal. With a 24V, 20-26 Ah (Plexus) or 10 Ah (Micro-Plexus) battery, the system can operate for 2-4 hours at maximum draw or 6-8 hours at nominal draw.

4.5 System Grounding

4.5.1 General Considerations

A good earth ground is necessary to ensure proper operation of the system. Carefully check that the system is connected to a reliable grounding path. Generally, the gauge of the ground wire should directly correlate with the distance from the ground source (i.e., a greater distance from the grounding source requires a larger gauge of grounding wire). The ground wire should be connected to the ground lug located inside the Plexus cabinet directly below the power supply. The distance of ground wire should be kept as short as possible.

4.5.2 Lightning Protection

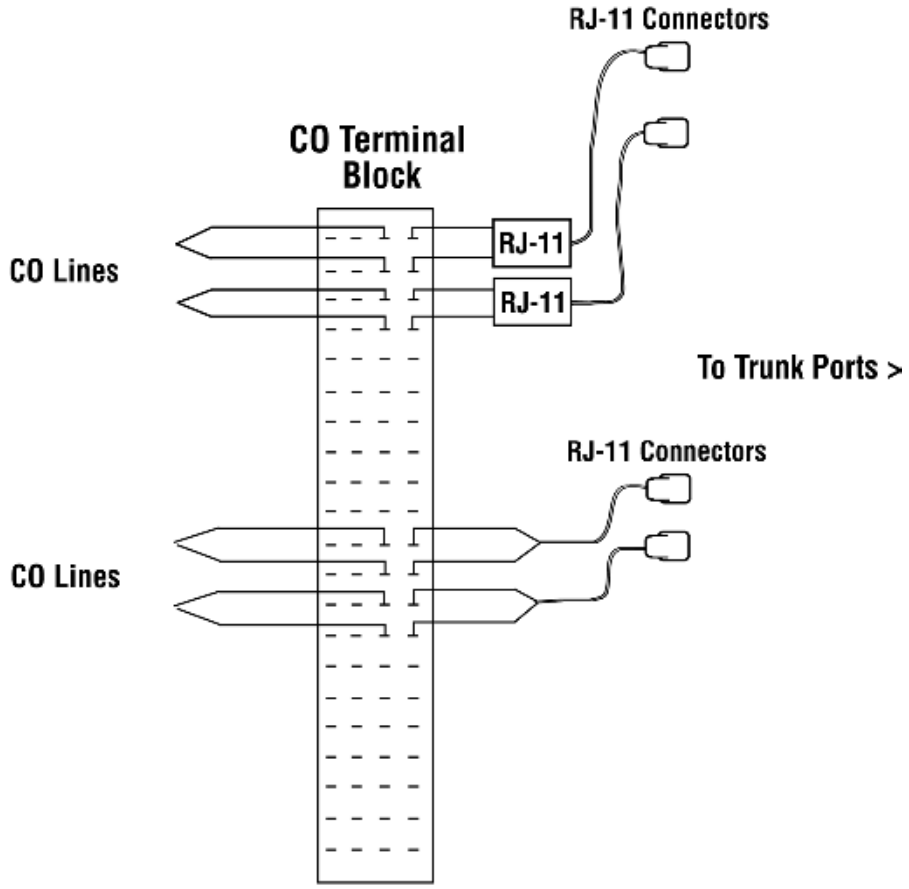
It is good practice to protect all CO lines and extension lines with proper lightning surge arrestors, such as gas discharge tubes. This guards against damaging surges caused by non-direct lightning strikes.

Failure to provide proper lightning protection may increase maintenance expense by requiring the replacement of damaged components.

4.6 Trunk Connections

4.6.1 Central Office (CO) Line Installation

Wire directly from the CO terminal block to the trunk ports of the Plexus system using only a two conductor (single pair) line cord from the terminal block, terminating with a RJ-11 plug. Depending on how the CO terminal block was installed, there may be modular jacks available to connect the two conductor (single pair) line cords between the trunk ports of the Plexus system and the CO terminal block.



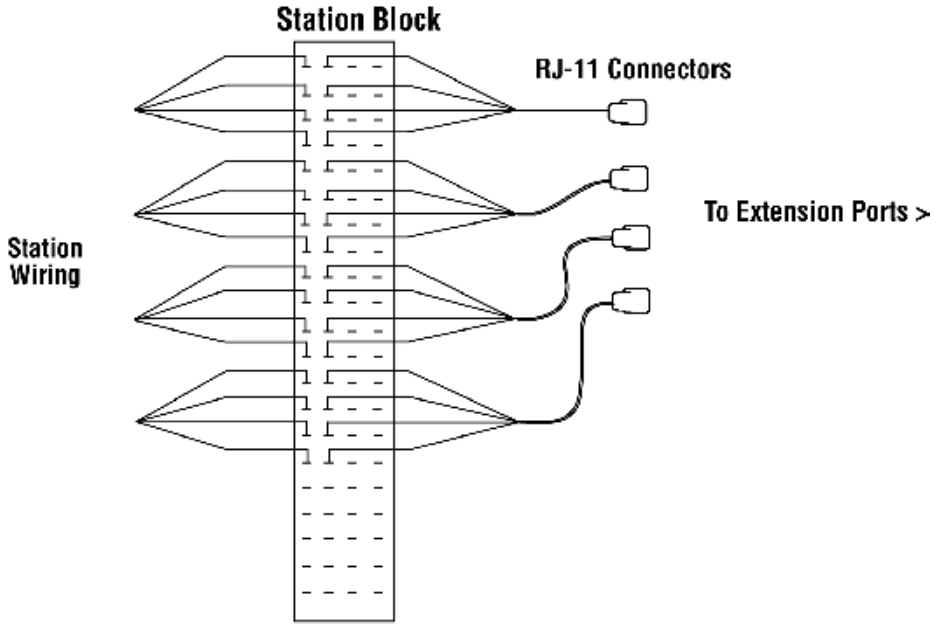
4.6.2 Surge Protection Considerations

Transitory voltage spikes, if induced onto CO lines, can travel through the cable and into the common equipment. The telephone company offers basic protection against this condition, but it is usually designed to protect the CO circuits and should, therefore, not be relied upon for total protection. To help protect the Plexus system from external voltage surges, the manufacturer recommends that gas discharge tubes, or similar primary protection devices, be installed and properly grounded on all lines.

4.7 Extension Connections

4.7.1 Extension Line Installation

Connections between the extension ports of the Plexus system and stations are typically done via punchdown blocks with four wire (two pair) line cords. The lines are terminated into the appropriate extension port with RJ-11 connectors.



For the Plexus system to function properly, the internal phone wiring must be in a “Home Run” configuration. If any two or more phones in a building access the same line when picked up, it is likely that the building is wired in a “Loop Through” configuration. In such a case, the wiring scheme must be redone in a “Home Run” configuration, as it will not function otherwise.

4.7.2 Distance Considerations

The maximum distance at which a station can be wired depends on the following two factors:

- The device being used at the station.
- The gauge of the wire being used for the extension (station) run.

The total resistance for the run should not exceed 700 Ohms. Therefore, if the device on the far end operates at a resistance of 350 Ohms, then a run of up to 1.5km using 24 AWG twisted pair copper wire would be acceptable.

Note
24 AWG copper wire provides approximately 180 Ohms of resistance per kilometer.

4.7.3 Grounding Considerations

If spare conductors exist in the cables run between the stations and the terminal block, it is good practice to connect them to earth ground. This action may prevent them form introducing Radio-Frequency Interference (RFI) and/or AC interference into the system. It is also good practice to disconnect any unused station jacks from the connector block and ground the wiring.

4.7.4 Station Relocation

The station relocation feature allows a station to be added, moved, or otherwise changed without relocating station wiring. Station relocation is possible with the Plexus because the extension number is based on a logical numbering plan (User ID) as opposed to the location of physical wiring. Once the telephone type is defined for an extension port, a user may log in at the station, enabling their unique system rights and privileges.

Note that some features, manually programmed into the key telephone, are not disabled or changed through user login (e.g., Caller ID format, LCD contrast, etc.).

4.8 Plexus Key Telephone Installation

4.8.1 PVT series analog Key Telephones

The wiring scheme for the Plexus analog key telephones (PVT-30, PVT-30D, PVT-22, PVT-22D) calls for straight-through wiring, using all four wires (two pairs).

Note
Be careful not to reverse the polarity when wiring. Reversing the polarity will prevent proper functioning of the analog key telephone.

A 4-wire base cord is shipped with each Plexus PVT analog key telephone. Use of this cord is recommended.

If a Plexus analog key telephone is not functioning properly when connected to the station wiring, perform the following check:

1. Connect the Plexus analog key telephone directly to the modular jack on the Analog Extension Interface (AEI) card using the 4-wire base cord shipped with the telephone.
2. If the telephone functions correctly, it is likely that the polarity is reversed in the station wiring, there is a wiring short, or there is a loose connection.
3. If the telephone does *not* function correctly, it is likely that the telephone or the AEI peripheral card is damaged.

Note
If the outer pair (data pair) on any wiring attached to an analog port on the Plexus system should short out, all data pairs on that card will cease operating until the shorted out pair is removed or un-shorted and the Plexus system is reset.

4.8.2 PDT series digital Key Telephones

The wiring scheme for the Plexus digital key telephones (PDT-30D and PDT-22D) calls for wiring, using two wires (single pair). Polarity is not an issue with Plexus digital key telephones.

A base cord is shipped with each Plexus PDT digital key telephone. Use of this cord is recommended.

If a Plexus digital key telephone is not functioning properly when connected to the station wiring, perform the following check:

1. Connect the Plexus digital key telephone directly to the modular jack on the Digital Extension Interface (DEI) card using the base cord shipped with the telephone.
2. If the telephone does function correctly, it is likely that the wiring has a short or loose connection.
3. If the telephone does *not* function correctly, it is likely that the telephone or the DEI peripheral card is damaged.

4.9 Analog Telephone/Device Installation

Ordinary phones and single-line telephone devices are wired using two conductor (single pair) base cords between the wall jack and the telephone set. Four wire (two pair) base cords should *not* be utilized with any ordinary phone or single-line telephone device. Such wiring may cause a short in the Plexus system. Utilizing four wire (two pair) wiring from the extension port to the station wall jack is not recommended due to the risk of the outer pair shorting out.

Note
If the outer pair (data pair) on any wiring attached to an analog port on the Plexus system should short out, all data pairs on that card will cease operating.

If a Plexus analog key telephone was previously connected to an extension port, it will be necessary to reset the port as follows:

1. After connecting the telephone/device, enter **F99** (default **F** = * on the keypad).
2. The system will reinitialize the port and identify the new device.

4.10 Operator Console Installation

4.10.1 DSS-60 Analog Operator Console

The wiring scheme for the Plexus analog operator consoles (DSS-60) calls for straight-through wiring using all four wires (two pairs). The polarity is *not* reversed. Note that reversing the polarity will prevent proper functioning of the operator console.

A base cord is shipped with each Plexus analog operator console. Use of this cord is recommended.

Note
Plexus analog operator consoles must be terminated on the same Analog Extension Interface (AEI) card as the host analog key telephone.

4.10.1.1 Installation Considerations

The DSS-60 Analog Operator Console is installed in the same manner as any other extension device (e.g., Plexus key telephone, ordinary phone, etc.). Each DSS-60 utilizes an extension port on an Analog Extension Interface (AEI) card. Therefore, separate station wiring must be available for each Operator Console.

To facilitate programming, each DSS-60 must have a host Plexus analog key telephone (i.e., PVT-30D). Each host is capable of programming up to seven (7) DSS-60 Analog Operator Consoles. All of the DSS-60 Analog Operator Consoles must be connected to the same AEI card as the host analog key telephone.

Note
The DSS-60 analog console only operates with the Plexus Analog Key Telephone of the same software version number (e.g., 1.01 or 2.0).

4.10.2 DDSS-60 Digital Operator Console

The wiring scheme for the Plexus digital operator consoles (DDSS-60) calls for wiring using two wires (single pair). The wiring polarity does not affect the DDSS-60.

A base cord is shipped with each Plexus digital operator console. Use of this cord is recommended.

Note
Plexus digital operator consoles must be terminated on the same Digital Extension Interface (DEI) card as the host digital key telephone.

4.10.2.1 Installation Considerations

The DDSS-60 Digital Operator Console is installed in the same manner as any other extension device (e.g., Plexus digital key telephone). Each DDSS-60 utilizes an extension port on a Digital Extension Interface (DEI) card. Therefore, separate station wiring must be available for each Operator Console.

To facilitate programming, each DDSS-60 must have a host Plexus digital key telephone (i.e., PDT-30D). Each host is capable of programming up to seven (7) DDSS-60 Digital Operator Consoles. All of the DDSS-60 Digital Operator Consoles must be connected to the same DEI card as the host digital key telephone.

Note
The DDSS-60 digital operator console only operates with the Plexus Digital Key Telephone.

4.10.3 Configuration Considerations

Operator Consoles should be assigned sequential User IDs based on the User ID (i.e., extension number) of the host key telephone.

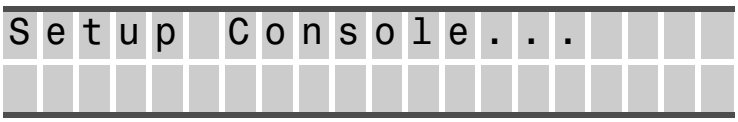
If the host key telephone is assigned a User ID of 100, the related DSS-60/DDSS-60 Operator Console(s) should have sequential User IDs beginning with 101.

4.10.4 Programming the Operator Console(s)

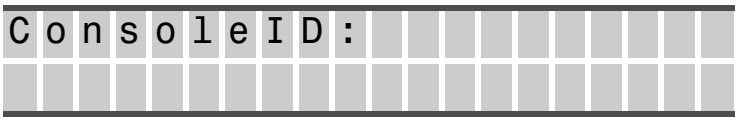
Once installed and configured, the DSS-60/DDSS-60 must be programmed using the host key telephone. It is recommended that each extension have an assigned key on the DSS-60/DDSS-60 Operator Console(s). These keys allow the operator to monitor phone activity and transfer calls easily.

4.10.4.1 Assigning Extension Keys

1. On the host key telephone, press and hold down **OVERRIDE** for 2 seconds.
2. Press **VOL** ▲ until the option **Setup Console** appears.



3. Press **SPKR** to select this mode.
4. At the **Console ID** prompt enter the User ID that has been assigned to the desired Operator Console.



5. Press **SPKR**.

All the programmable keys on the corresponding Operator Console will illuminate.

6 On the Operator Console, press the Operator Console key to be programmed.

7 On the host key telephone, press **VOL▲** until the desired parameter appears.

```
Monitor
ID:
```

8 Operator Consoles can have each key used to monitoring extensions or trunks, provide access to features or functions, or as filter keys for special applications (e.g., Hotel and Motel).

Note
For Information on feature or function codes, see chapter 8, "Feature Reference". For information on filter codes, see chapter 18, "Hotel / Motel Package".

9 Enter the appropriate system entity ID, feature or function code, or filter code.

10 Press the next key to program.

11 Repeat steps the above steps until each key has a defined setting.

12 Press **OVERRIDE** to exit out of the programming mode.

Tip
Label each key on the DSS-60/DDSS-60 with the appropriate system entity number or code. Precut overlay templates and software to customize the labels are available from dealers or Customer Service.

4.11 External Paging (Pager) Interface

4.11.1 Installation Considerations

The Plexus system provides a special mono RCA line-out connector (labeled "Pager") which may be used to couple the system to an external paging amplifier (not included). The line-out port does not provide a talk-back path nor will it recognize DTMF tones.

Note
Use of either the paging or the music-on-hold interface (or both) on a Micro system utilizes the resources of extension port #8 on the motherboard.

Warning
Some paging systems or music devices put out large static voltage which could damage the circuitry in the Plexus system. Install equipment with the following safeguards.

You will need to protect the Plexus systems against excessive stray voltage output. The design below provides for two capacitors to prevent stray voltage from the pager's amplifier or from music systems from reaching and damaging Plexus circuitry. Resisters are included in the circuit to dampen any surge effects. All parts can be found at any electronics retailer.

A. Materials

Components	Quantity needed
1K, 1/2 W, 5% resistor	2
0.22 MF, 50 V, disc capacitor	2
RCA Plug	1
red and green wire	as needed

B. Directions

1. Crimp or solder the red wire to the pole (or tip) contact of the RCA connector.
2. Crimp or solder the green wire to the shell (or ring) contact of the RCA connector.

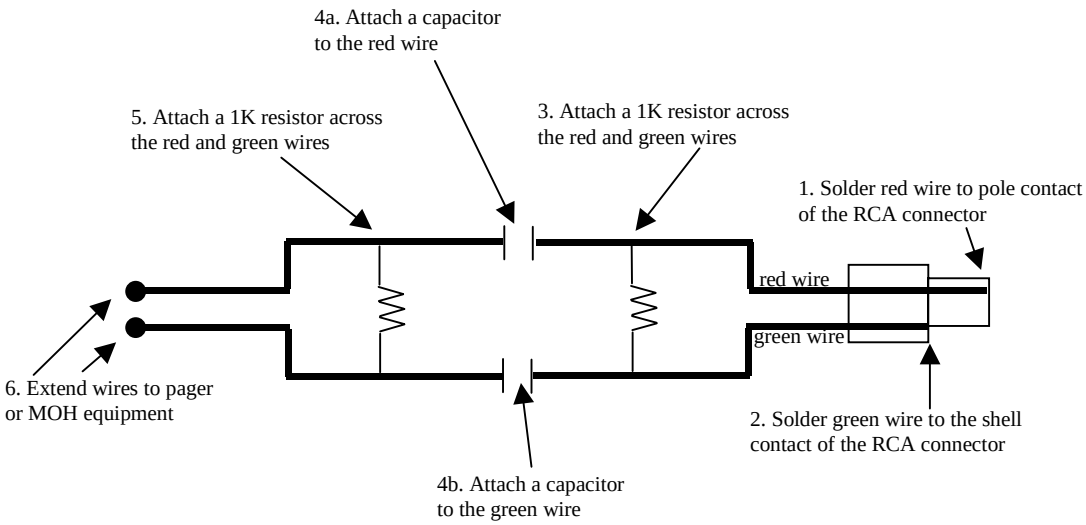
3 Attach a 1K resistor across the red and green wires.

4 Attach a capacitor to the red wire and green wire, as shown.

5 Attach a second 1K resistor across the red and green wires.

6 Extend wires to the pager or music on hold equipment.

7 Plug the RCA jack into the PAGER or MOH port.



4.12 Music-on-hold interface

4.12.1 Installation Considerations

The Plexus system provides a special mono RCA line-in connector (labeled "MOH") which may be used in connecting a music source (not included) for Music-On-Hold. The system supplies the signal from the music source to outside lines placed on hold.

Note
Use of either the music-on-hold or the paging interface (or both) on a Micro system utilizes the resources of extension port #8 on the motherboard.

Warning
Some paging systems or music devices put out large static voltage which could damage the circuitry in the Plexus system. Install such devices according to directions in previous section, External Paging (Pager) Interface.

4.13 Data Communication Interface

4.13.1 Installation Considerations

The Plexus system is connected to a computer through the port labeled “RS-232”. A straight (*not* null modem) 9-pin serial cable is connected between this port and an available serial port on the computer.

All data transfers to and from the Plexus system are performed according to the following communications rate and format:

- 19200 baud
- 8 data bits
- 1 stop bit
- no parity

4.13.1.1 RS-232 Pin-Out

The pin-out for the RS-232 connection is as follows:

Pin	Function
1	No Connection
2	Data out of KSU
3	Data into KSU
4	Data Terminal Ready
5	Ground
6	Data Set Ready
7	Request to Send
8	Clear to Send
9	No Connection

4.14 System Check-Out

System operating features are set to defaults at initial power-up (prior to any configuration). The default settings provide a basic operating system with a known set of parameters. Initial system checks should be performed under these conditions.

4.14.1 General Check

The red LED located on the face of the DXP card is the system status indicator. When power is applied to the Plexus system, the LED should flash steadily at the rate of 1 second on and 1 second off.

Different flashing rates or the lack of flashing indicates problems as follows:

- An LED that remains lit indicates that the system has crashed.
- Rapid flashing indicates that the system firmware must be reloaded.
- Steady flashing for a period of 2 seconds on and 2 seconds off indicates a hardware fault.
- Steady flashing for a period of 4 seconds on and 4 seconds off indicates a possible DSP problem.

The common equipment and telephone installation may be checked for proper operation by performing the resistance and voltage checks as discussed below.

4.14.2 Resistance Check

The resistance of each installed station as measured from the station side of the station connector block(s) should be within the following limits:

Greater than 700 KOhms

Note
Resistance values vary with cable length and station type.

Measure resistance at the station connector block(s) under the following conditions:

- Plexus AC power cord disconnected from electrical outlet

- Hybrid-key extension ports wired to station connector block(s)
- Stations wired with wiring punched down at station connector block(s)
- Bridging clips removed from station connector block(s) to isolate stations from common equipment

4.14.3 Voltage Check

The voltage as measured across the signal pair (or inner pair) at the station connector block(s) must be within the following limits:

28-36 VDC

Measure voltage at the station connector block(s) under the following conditions:

- Plexus AC power cord connected to an electrical outlet
- Bridging clips installed at the station connector block(s)

4.14.4 Direct Port Dialing Check

The Direct Port Dialing Check is a simple test to isolate the failure of either an extension port or a trunk port.

To perform the check, proceed as follows:

4.14.4.1 Prior to Configuring the System

1. Insert the Analog Trunk Interface (ATI) and/or Analog Extension Interface (AEI) cards.
2. Power up the Plexus system.
3. Connect a Plexus analog key telephone or an ordinary phone to any analog extension port.
4. Connect a Plexus analog key telephone or an ordinary phone to the analog *extension port* to be checked.

   or

   Connect a base cord between the *trunk port* to be checked and a previously checked extension port.
5. From the phone connected in Step 3, enter # + Unit ID, Slot ID, and Port ID of the port to be checked.

For example, to test the top port on an AEI card inserted into the first slot on a single unlinked Macro Plexus system, the following string would be entered:

 # 0 0 00

Note
All port IDs are two digits although the software may only show one digit. If only one digit is shown, add a 0 to the beginning of the Port ID.

4.14.4.2 Subsequent to Configuration

When testing a port on a system that has already been configured, System Administrator Privileges must be activated prior to entering the test string on the phone. In such cases, Step 4 would change as follows:

1. From the phone connected in Step 3, enter **F91** + Password. The Default Password is **123456**.
2. Enter **#** + Unit ID, Slot ID, and Port ID of the port to be checked.

5

Plexus Configuration Wizard

5.1 Introduction to Plexus Configuration Wizard

Welcome to the Plexus Configuration Wizard. If you are learning about the system for the first time or programming a brand new system, the Wizard will automatically program the system with a basic configuration.

For new users, the Plexus Configuration Wizard can set up a functioning configuration. This configuration will set up all parameters to allow basic access to features, trunks, extensions, voice-mail, and auto-attendants.

For experienced users, the Plexus Configuration Wizard provides a major time savings. The configuration, including voice mailboxes and a basic auto attendant, is created with minimal effort. This configuration ensures that all parameters have been set to provide access to features, trunks, extensions, voice-mail, and auto-attendants. Another benefit is that all new configurations will be set up with a similar structure, aiding in system maintenance.

5.2 About Plexus Configuration Wizard

The Plexus Configuration Wizard offers a “jump-start” facility to create a basic configuration. The configuration allows the core functions relating to the trunks, extensions, voice-mail, and auto-attendants, to be set up quick and easily to provide a base level of operationality. For many systems, additional modifications may be necessary.

This facility will **NOT** provide access to all the options, interfaces or advanced features within the system’s capabilities. To access advanced features or to modify the configuration after it has been created by the Plexus Configuration Wizard, see “Software configuration - with Plexus Configuration Wizard” - section 6.2.

Note
Use the Plexus Configuration Wizard for new configurations or installations only. The Plexus Configuration Wizard will clear and reset any configuration currently in a system.

The Plexus Configuration Wizard creates a configuration with the number of CO lines specified and extensions for each extension port available on the system. An operator user group (user group 300) will be created with user 100 a member of that user group. If the system has an Integrated Voice Processor (IVP) card installed, the Plexus Configuration Wizard will create mailboxes for all the users and the user group and a basic auto attendant. See “Configuration Report” - section 5.4 and read the example configuration report for information on other parameters created by the Plexus Configuration Wizard.

5.3 Using the Configuration Wizard

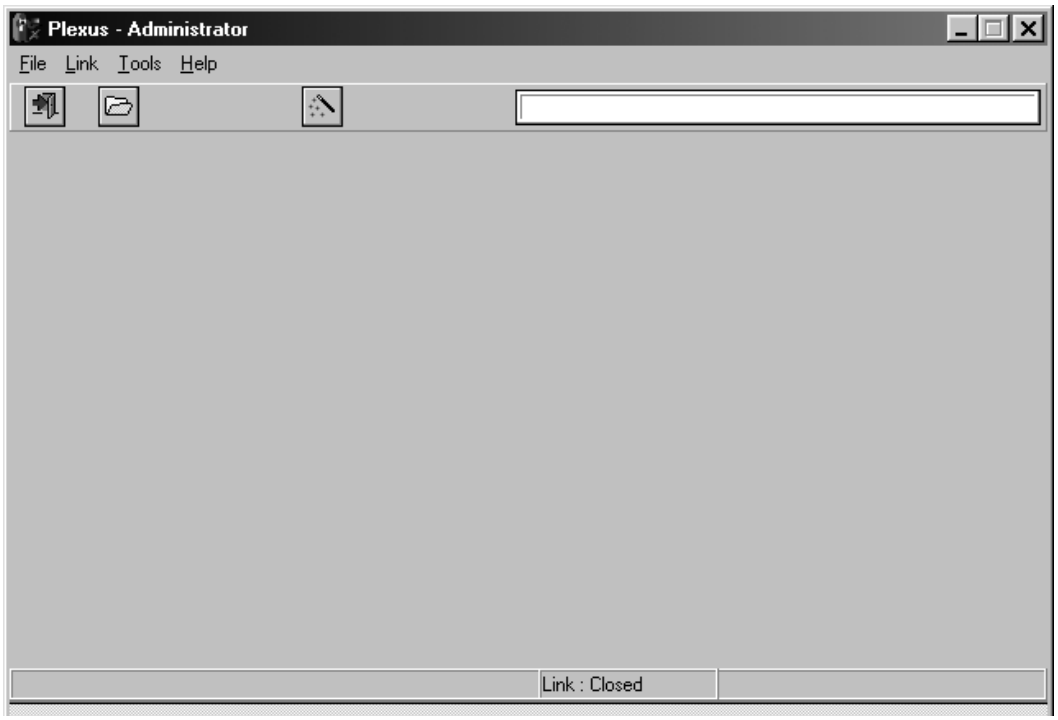
For information on installing and setting up the Plexus Administrator software see “Install Plexus Administrator” - section 3.2.

Perform the following steps to use the Plexus Configuration Wizard.

Note
The example given is for a Micro Plexus with 4 CO lines and 8 extensions. Different models or configurations will produce slightly different windows.

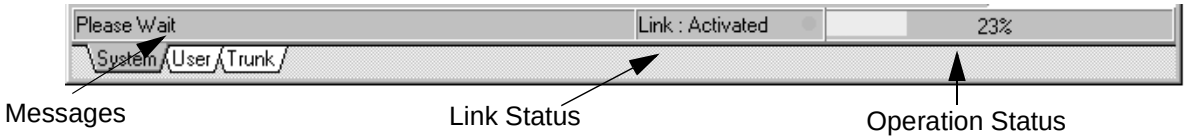
1 Start the Plexus Administrator.

The following screen is displayed.



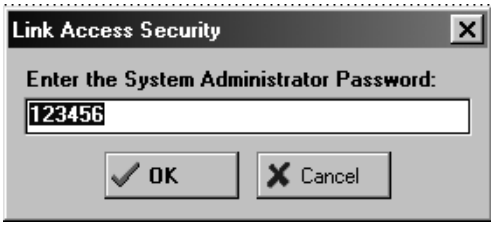
The bottom of Plexus Administrator Window contains a status bar that provides:

- Messages
- Link Status
- Operation Completion Status



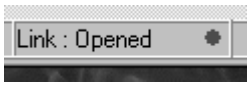
2 Establish a link between the Plexus Administrator and the Plexus System by selecting Open on the Link pull-down menu.

The Link Access Security window is displayed



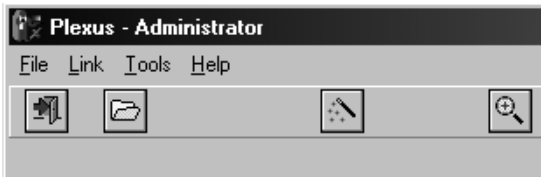
3 Enter the System Administrator Password (the default is 123456) and click **OK**.

The Link Indicator in the status bar of the Administrator Window indicates that the link has been opened.



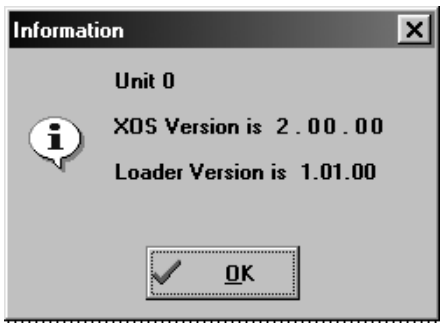
Once the link is successfully activated the indicator changes color and

an magnifying glass icon appears in the Plexus Window.



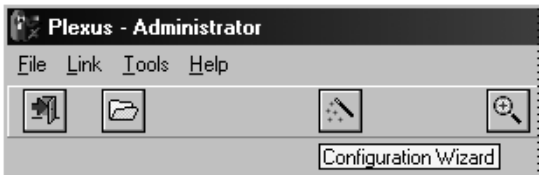
4 Click on the magnifying glass icon. This verifies that the link with the system is valid and also verifies that the Plexus system has a compatible XOS version by querying the XOS version of the system and receiving a response. The link has been successfully activated if a pop-up window with the XOS version appears.

Note
If a message appears "This version of Plexus Administrator is not Compatible with the current version of XOS." you must upgrade the XOS version of the Plexus system before using the Plexus Configuration Wizard. See “Upgrading version 1.02 XOS with a version of 2.0 XOS” - section 3.3.2.

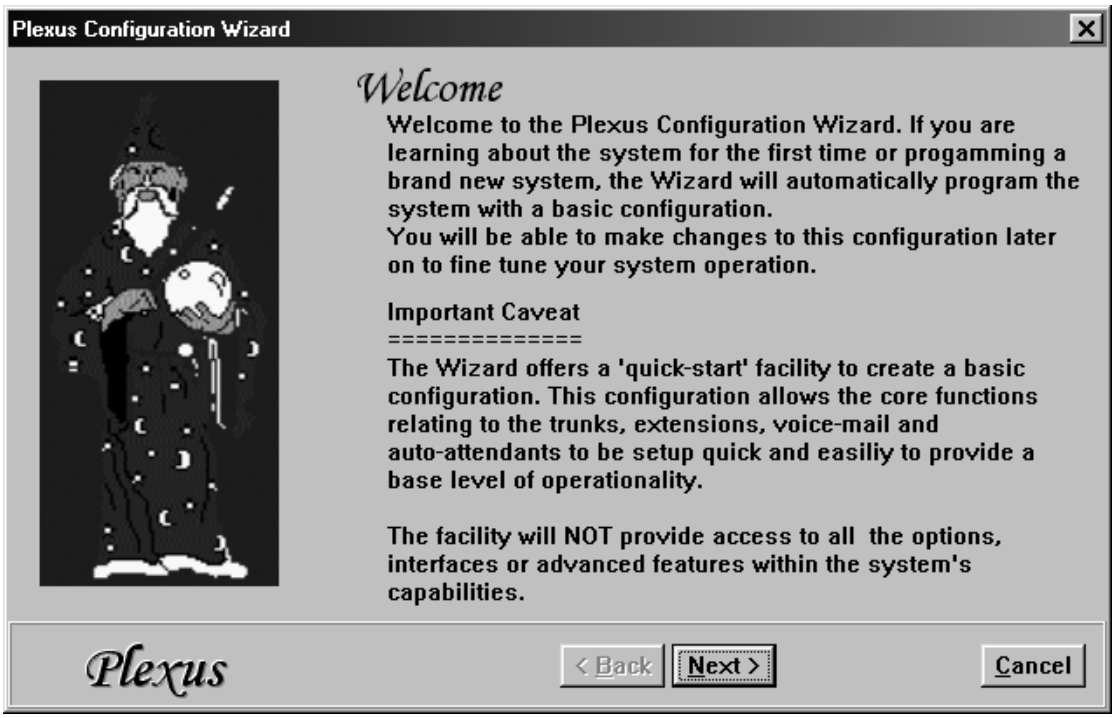


5 Click **OK**.

6 Click the Configuration Wizard button on the Administrator Icon Menu Bar.

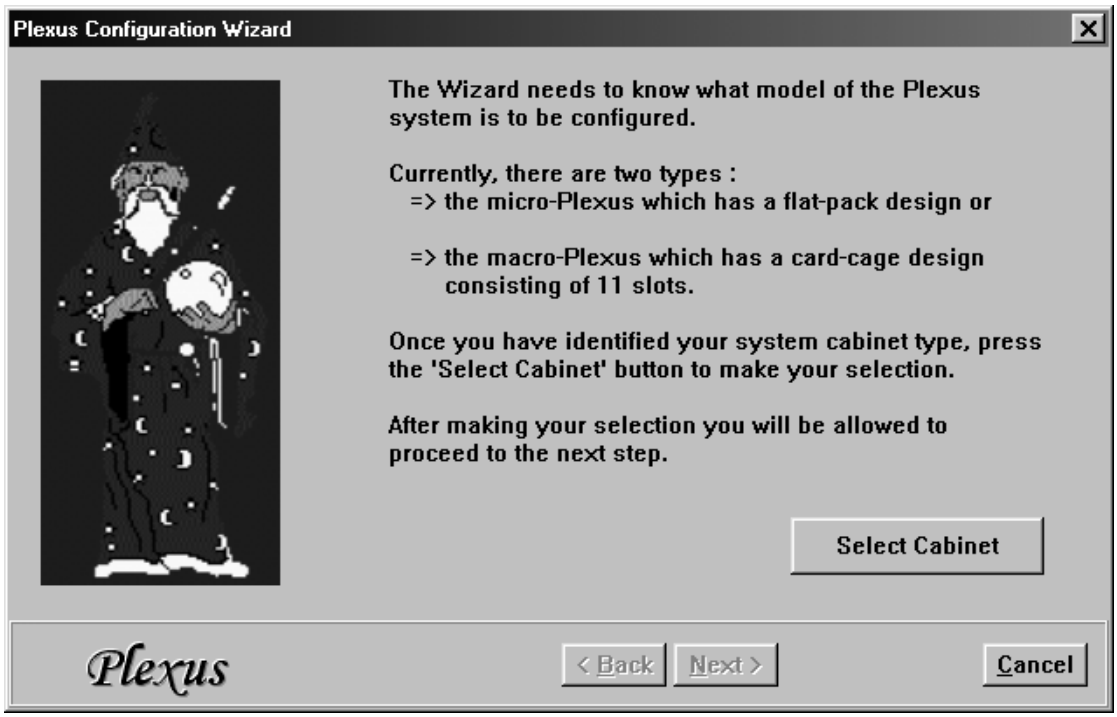


This displays the "Welcome" screen of the Plexus Configuration Wizard



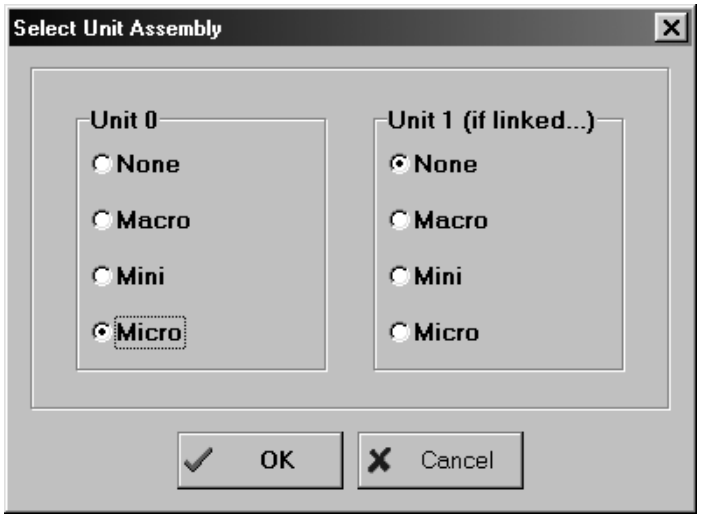
7 Click on the Next button to continue.

8 Select the type of system you are installing, either macro-Plexus or micro-Plexus, by clicking the Select Cabinet button.



Clicking on the Select Cabinet button causes the "Select Unit Assembly" window to display.

.

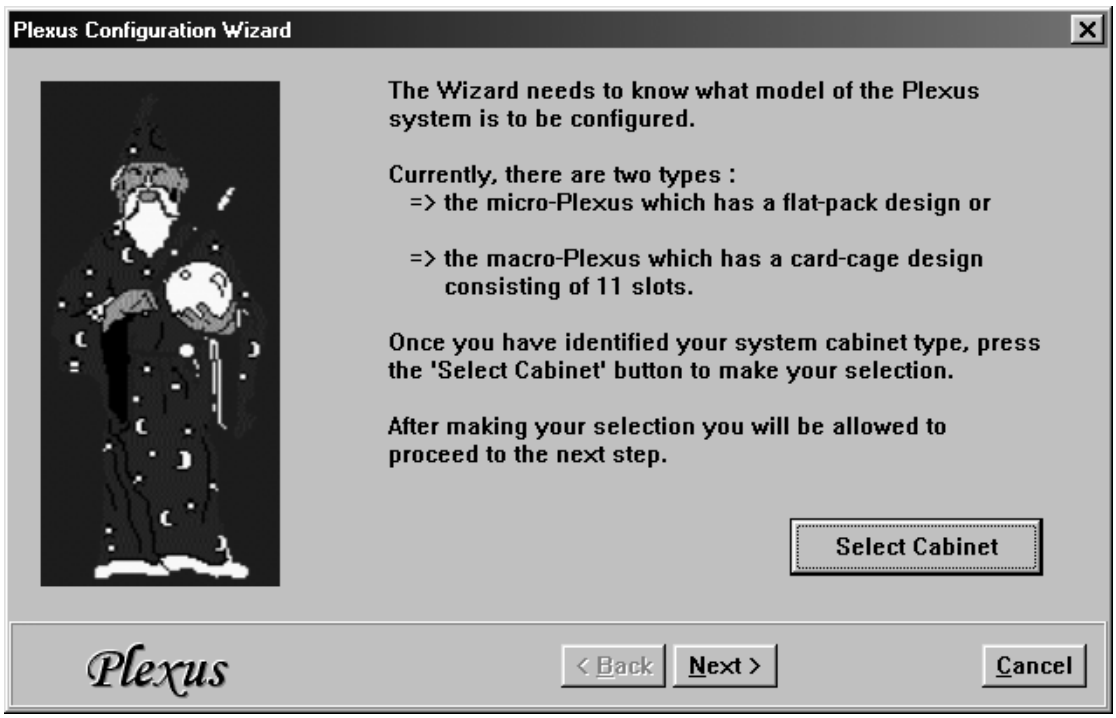


9 Click on the type on Plexus you are installing.

Note
If you are installing two linked Plexus Systems, you must specify one unit for each Plexus you are installing.

10 Click OK after selecting the units you are installing.

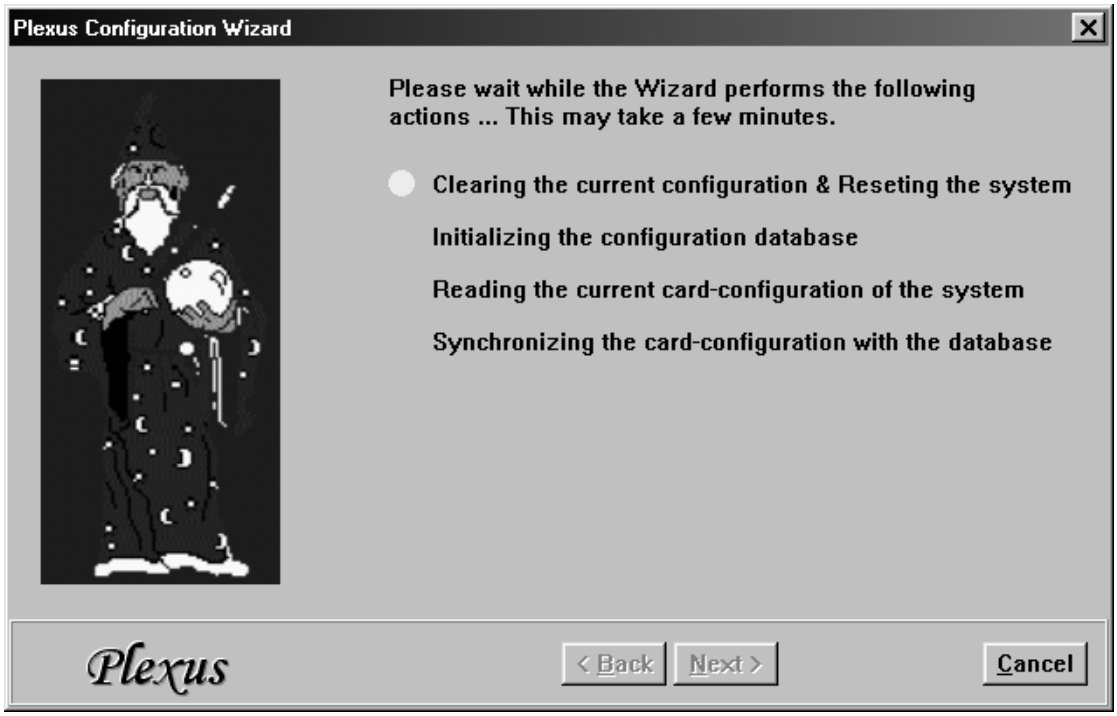
Clicking OK brings the last window back up but with the NEXT button highlighted



Clicking Next enables the Configuration Wizard to

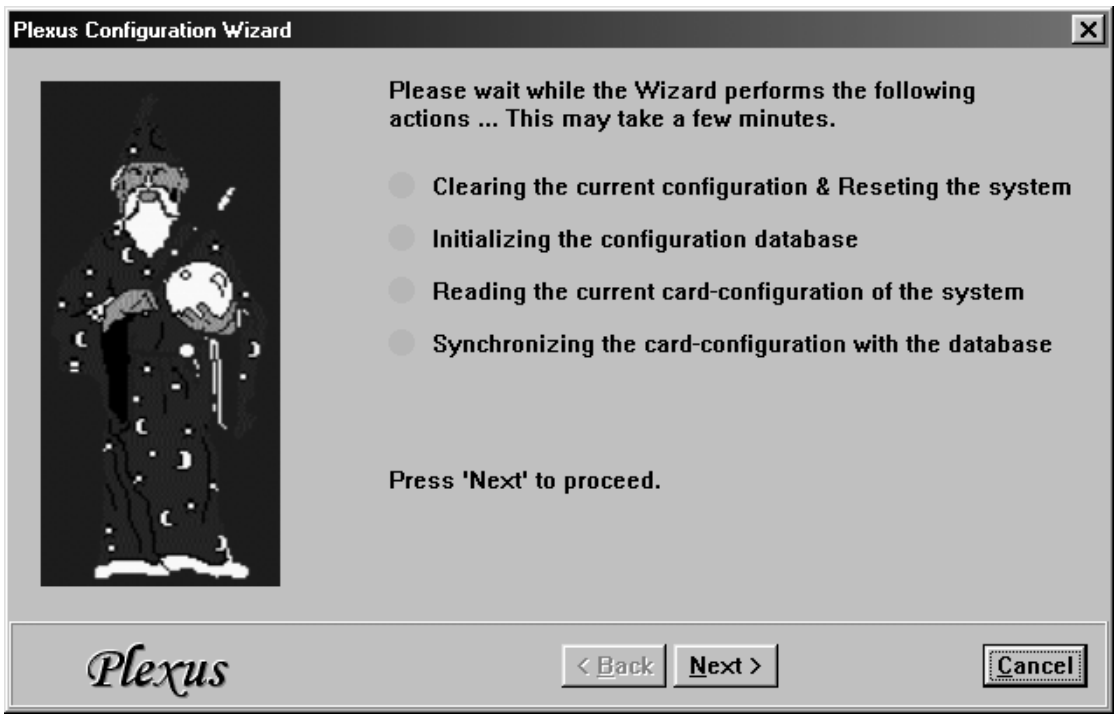
- Clear and reset the system
- Initialize the configuration database
- Read the current card configuration
- Synchronize the card configuration with the database

The steps will be tracked by the Plexus Configuration Wizard.



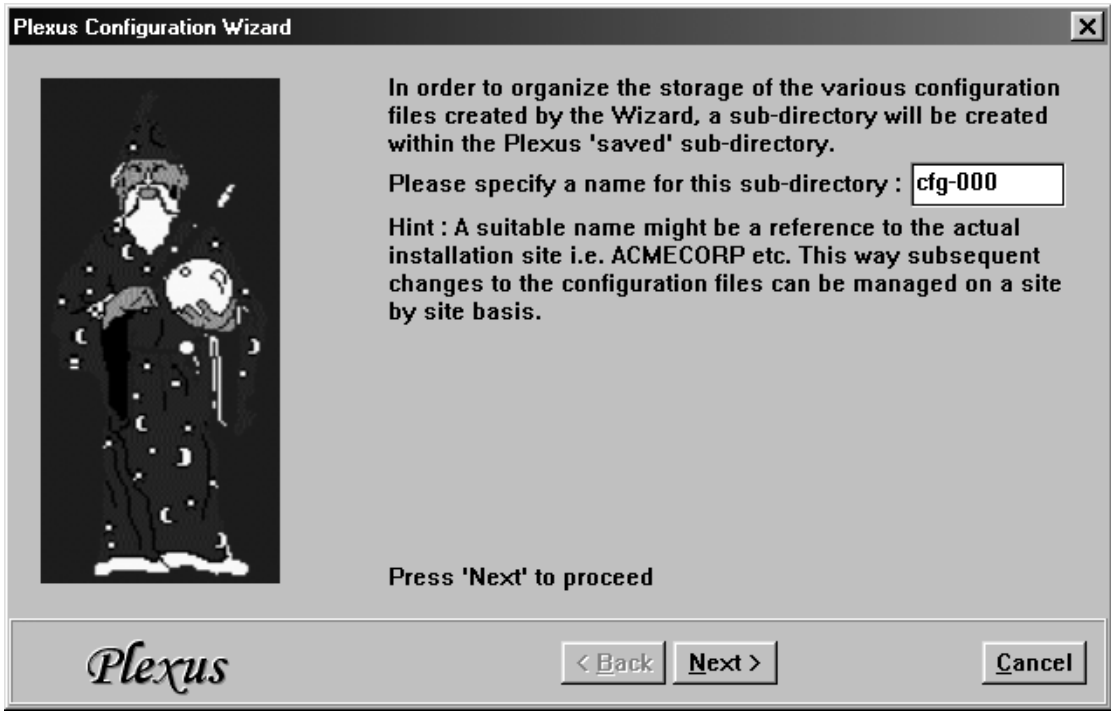
As the Wizard completes these actions, buttons display indicating what is being done and its status. (Yellow = In-Progress, Green = Complete,)

When the configuration has completed these actions, the following screen is displayed.



11 Click Next to continue.

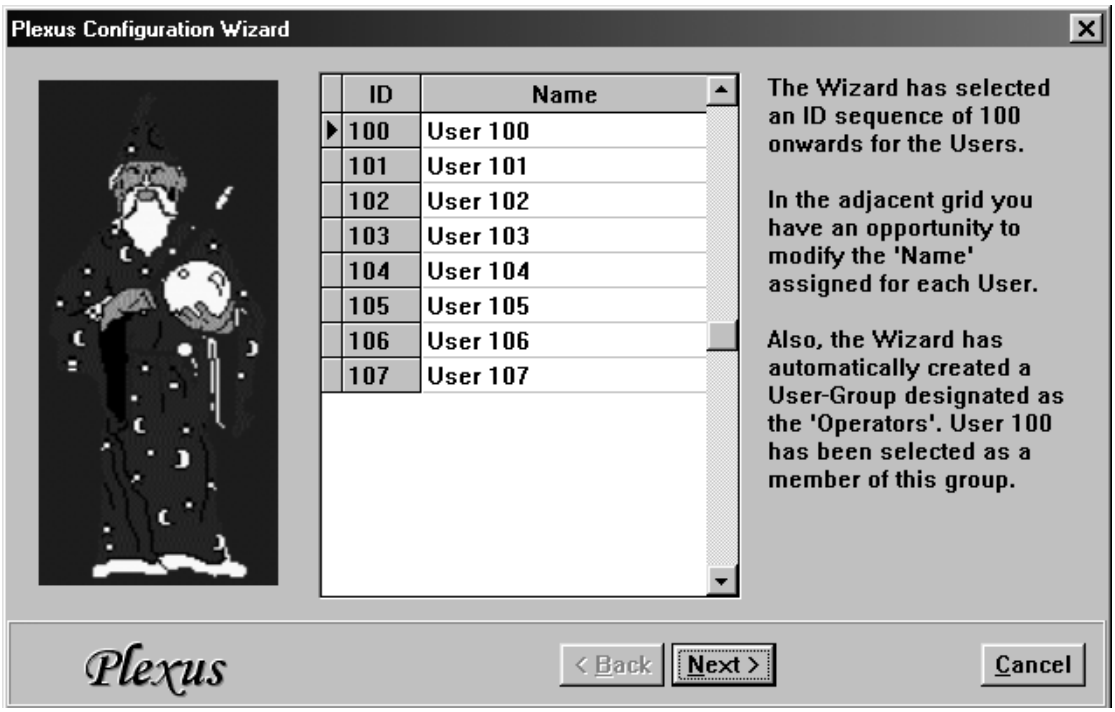
The Wizard now displays the sub-directory screen.



12 Specify a unique name, if desired, for the sub-directory to which the Wizard will save files. The Wizard will give the default name **cfg-000**.

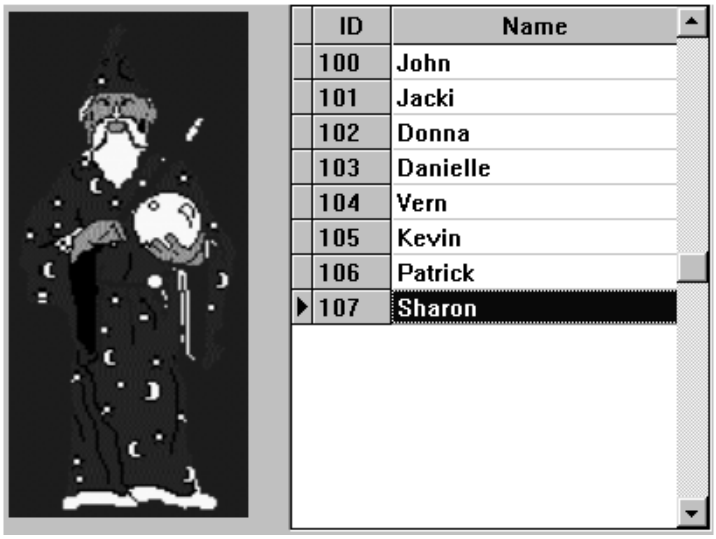
13 Click Next to continue.

The Wizard now displays the user entry screen.



This screen allows you to enter the individual names assigned to each extension of your system.

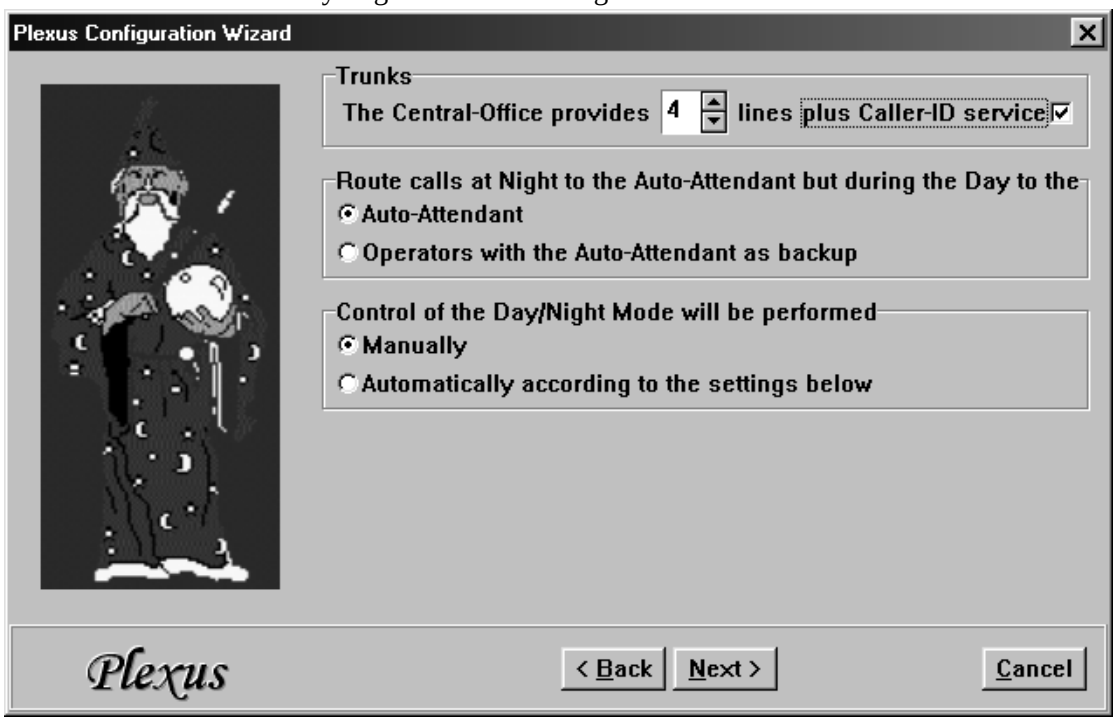
14 Enter the name of the person assigned to each extension.



15 Click Next to continue.

The next screen in the Wizard allows you to configure the following options:

- The number of trunks and whether Caller ID is supplied.
- Whether to route calls during the day through the Auto-Attendant
- Whether the Day/Night Mode switching is automated or manual.



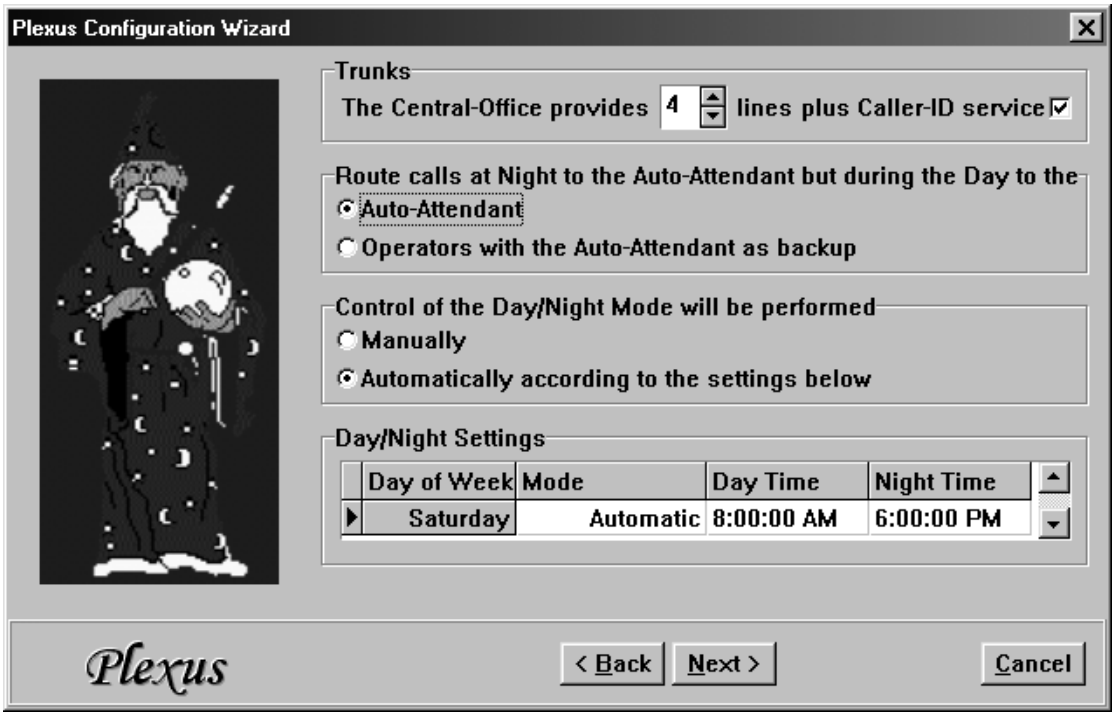
16 Select the number of trunks and indicate whether Caller ID service is available.

17 Select how the calls will be routed during the "Day" by clicking either "Auto Attendant" or Operators with the Auto Attendant as backup.

Select whether the Day/Night switching is done "Manually" or "Automatically" according to the settings below.

Note
The default for Control of the Day/Night Mode is manual. This setting allows for flexibility in determining the open versus closed hours. The Plexus system provides a easy to use "one-touch" switching from day to night mode (e.g., an operator can switch into the night mode to route calls to a night auto attendant whenever leaving the office at night, no matter what the time). See section 8.63, "Day / Night Service" for more information.

Selecting the "Automatically according to the settings below" option causes the "Day/Night Settings Panel" to be displayed.



18 If the "Automatically according to the settings below" option was chosen, enter the "Day Time" for switching from night to day and enter the "Night Time" for switching from day to night. Use the arrows on the right to access different days. The table below shows the system time default settings.

Day of Week	Mode	Day Time	Night Time
Sunday	Automatic	8:00:00 AM	6:00:00 PM
Monday	Automatic	8:00:00 AM	6:00:00 PM
Tuesday	Automatic	8:00:00 AM	6:00:00 PM
Wednesday	Automatic	8:00:00 AM	6:00:00 PM
Thursday	Automatic	8:00:00 AM	6:00:00 PM
Friday	Automatic	8:00:00 AM	6:00:00 PM
Saturday	Automatic	8:00:00 AM	6:00:00 PM

19 Click Next to continue.

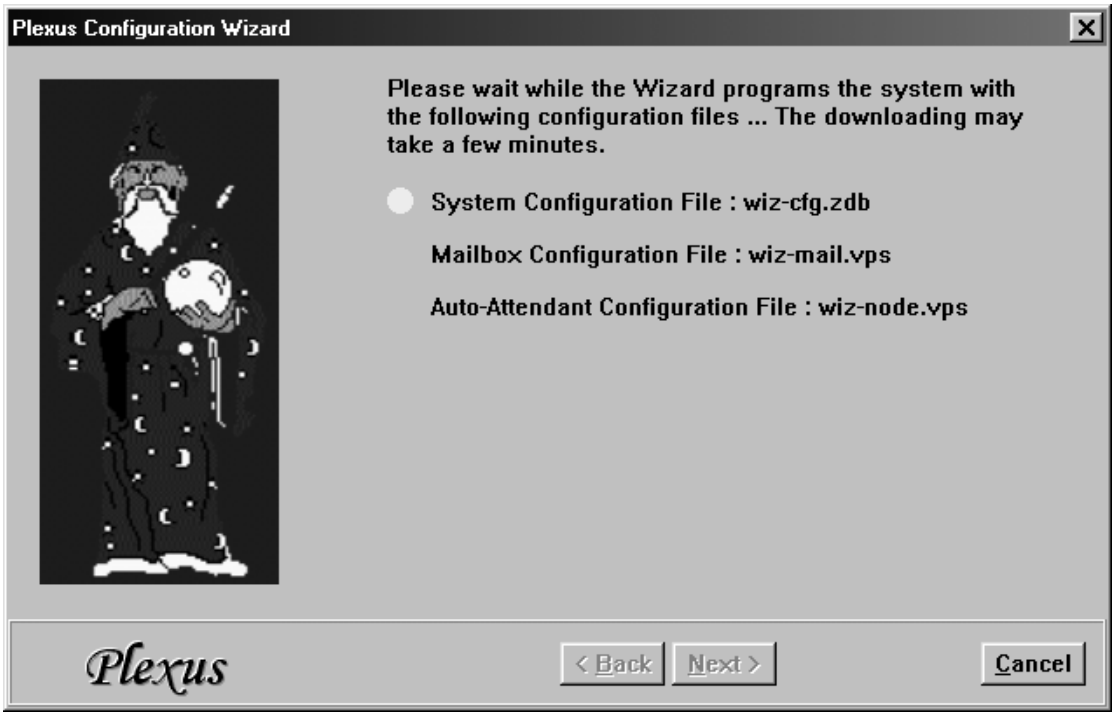
The Wizard is now building your Plexus System Configuration files. These include:

- System Configuration Files
- Mail Box Database
- Auto Attendant Database
- Mail Box Download Format File
- Auto Attendant Download Format File

20 When these files have been built, indicated by green buttons, click Next to continue.

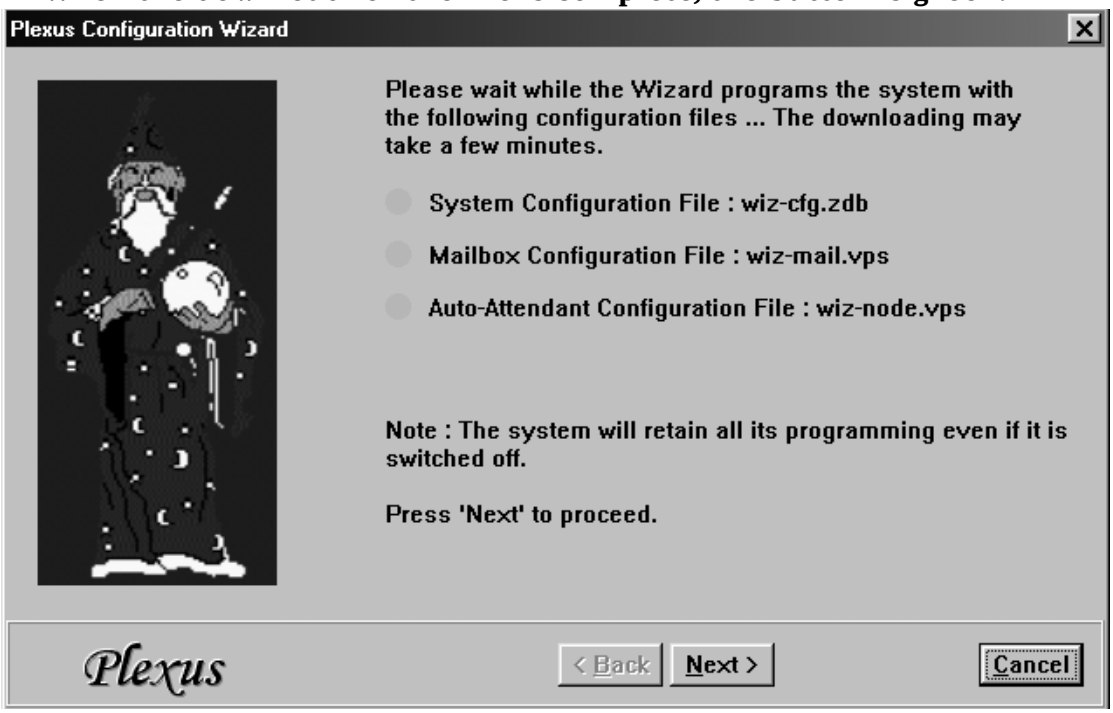
Note
If you choose the "Skip the download of the VPS files" option, the auto attendant structure and the voice mailboxes will not be uploaded into the system.

Clicking Next causes the configuration files to be downloaded to your system.



While each file is being downloaded, a yellow indicator button will be displayed.

When the download for the file is complete, the button is green.



21 After the files have been downloaded, click Next to continue.

22 Your installation is now complete. You may view a configuration file by clicking the “Show Configuration Report” button (see “Configuration Report” - section 5.4) or click “Finish” to exit the Wizard.

23 If modifications are necessary for the configuration, see “Software configuration - with Plexus Configuration Wizard” - section 6.2.

5.4 Configuration Report

A Configuration Report is created each time the Plexus Configuration Wizard creates a new configuration. The file is named **wiz-rep.txt** and is saved in the sub-directory specified at the beginning of the Plexus Configuration Wizard process. It includes several important pieces of information about the created configuration. These include:

- Date and time the configuration was created.
- User numbers and names
- User groups and members
- Trunks and trunk groups
- Auto attendant information
- Call routing information
- Voice mailbox information
- Port assignments
- A map of port assignments

The following is an example configuration report:

```
=================================================================
Plexus Wizard Configuration Report - created: 8/2/99 5:04:00 PM
=================================================================

Important Caveat
----------------
This basic configuration will allow the core functions relating
to the trunks, extensions, voice-mail (if present) and
auto-attendants (if present) to be setup quickly and easily to
provide a base level of operation.

The configuration will NOT contain the settings to all the options,
interfaces or advanced features within the system's capabilities.
To exact the full functionality of the system and thereby provide
the desired level of operation, it would require a thorough
understanding of the systems capabilities and be proficient in
using the configuration utility - the Plexus Administrator - to
properly program the system.
```

Please familiarize yourself with all the documentation provided to gain the full benefit of the system's capabilities.

Users & User-Groups
==========
The following Users have been created.
ID:100 Name: John
ID:101 Name: Jacki
ID:102 Name: Donna
ID:103 Name: Danielle
ID:104 Name: Vern
ID:105 Name: Kevin
ID:106 Name: Patrick
ID:107 Name: Sharon

Note: User 100 has been granted 'Administrator' privileges.

Refer to the documentation for all the other option settings and their defaults.

The following User-Group has been created.
ID: 300 Name: Operators

Note: User 100 has been included as a member of this group.

Trunks & Trunk-Groups
==========
The following Trunks have been created.
ID:201 Name: Trunk 201
ID:202 Name: Trunk 202
ID:203 Name: Trunk 203
ID:204 Name: Trunk 204

The following Trunk-Group has been created.
ID: 400 Name: CO-Lines

Note: All trunks have been included as members of this group.

Auto-Attendant
==========
A Day(ID:450) and Night(ID:451) Auto-Attendant has been created.

To customize the greetings, dial 450 or 451. During the announcement press <*>
Enter the System-Administrators mailbox: <005>, then the

password:<0000>.
Select auto-attendant option: <1>
To re-record the :-
- Day-Greeting : enter the modify recording option: <9>,
  followed by <00#>.
  Speak the greeting, and terminate with a <#>
  e.g. "Thank you for calling the ACME Corporation."

- Night-Greeting : enter the modify recording option: <9>,
  followed by <10#>.
  Speak the greeting, and terminate with a <#>
  e.g. "You have reached the ACME Corporation after hours.
       Our hours of operation are 9am to 5pm, Monday to
       Friday. Please call again."

- Menu-Prompt : enter the modify recording option: <9>,
  followed by <20#>.
  Speak the greeting, and terminate with a <#>
  e.g. "If you know your party's extension, you may dial it at
       anytime now. Otherwise, press zero or stay on the line
       for further assistance."

To test the setup just dial the appropriate Auto-Attendant ID
i.e. 450 - for the Day Auto-Attendant
     451 - for the Night Auto-Attendant

In-Coming Call Routing
======================
Control of the Day/Night mode will need to be performed manually.
This can be done from User 100's telephone using either :-
  Feature <*48> for Day Mode, and <*49> for Night Mode.
or
  If a Key-Telephone is present, defining a DSS Key with
  Function <04> and selecting the desired mode.
Please refer to the product documentation for further guidance.

All the trunks for in-coming calls have been routed to the Day
and Night Auto-Attendants as appropriate.

Voice-Mail
==========
Voice-Mailboxes have been created for all the users.
For the Operators - User-Group 300; a dispatch mailbox has been
created. User 100 is a member of the dispatch group.
All the mailboxes have a default password: <no password>

Port-Assignments
==========
The diagrams below provide a pictorial overview of the platforms and the installed interfaces. At each port location on the interfaces, the assigned User or Trunk is indicated by means of the User-ID or Trunk-ID as appropriate.

Use this as a guide for the final physical inter-connections of the extensions and CO-lines to the system once the wiring is in place.

Note: Port locations marked with an 'x' indicate that those ports are disabled and will not be operational. Do not connect any physical lines; neither extensions nor CO-lines to these ports. If this is necessary, the current configuration will have to be modified appropriately.

Important Note: Interfaces marked with '???' signify that it was not possible to fully configure these interfaces. This means that the resultant configuration may not be adequate for proper system operation. It would be wise to inspect the current configuration and make the necessary changes where appropriate. Please consult the documentation for specific information.

Platform: Micro - [Flat-Pack Chassis]
Unit: 0

Front : Lower Slot - Base

|ATI|201|202|203|204|-|100|101|102|103|104|105|106| x |AEI|

Front : Upper Slot - Expansion

| | | | | |-| | | | | | | | |

Rear : Upper Slot - Expansion

| | | | | |-| | | | | | | | |

==========

Software Configuration

6

6.1 Introduction

All aspects of the Plexus system can be configured using the Plexus Administrator Windows application.

About Software Configuration

A Plexus system can be configured using the Plexus Administrator two ways; using the Plexus Administrator with the Plexus Configuration Wizard or using the Plexus Administrator without using the Plexus Configuration Wizard.

Note
It is recommended that you create your configuration using the Plexus Configuration Wizard. The Plexus Configuration Wizard provides a valuable time savings. The configuration created allows the core functions relating to the trunks, extensions, voice-mail, and auto-attendants to be set up quick and easily to provide a base level of operation.

6.2 Software configuration - with Plexus Configuration Wizard

Several steps that must be completed in order to configure a Plexus system using the Plexus Administrator with the Plexus Configuration Wizard. The Wizard will create all the users and trunks and place them according to the physical layout of the system. You will be able to modify settings after the Plexus Configuration Wizard has created the configuration.

1. "Determine the structure of the system" - section 6.4.
2. "Physically insert peripheral cards into the cabinet" - section 6.5.
3. Follow the steps in Chapter 5, "Plexus Configuration Wizard". (The Wizard will create a working configuration. If modifications are necessary, continue to step 4.)
4. Launch Plexus Administrator if it is not already open (see "Launch Plexus Administrator" - section 6.6).
5. Open the created configuration (see "Using an existing configuration file" - section 6.8).
6. Go to "Configure Auto-Attendant Interface (AAI) card" - section 6.10. Follow all subsequent steps and sections to modify the configuration.

6.3 Software configuration - without Plexus Configuration Wizard

The outline that follows discusses the steps that must be completed in order to configure a Plexus system using the Plexus Administrator without using the Plexus Configuration Wizard. Several procedures and the various system entities (i.e. Users, User Groups, Trunks, Trunk Groups, Extension Ports, and Trunk Ports) are addressed. While all steps must be completed in order to successfully configure a system, the user does not need to complete each step according to the sequence shown below. The Plexus Administrator is fully modular and allows the user to address each parameter in any order. However there are certain dependencies e.g., create a user before configuring the user's coverage. This Software Configuration section is organized according to the following sequence. This sequence is followed in order through the rest of this chapter.

1 "Determine the structure of the system" - section 6.4

2 "Physically insert peripheral cards into the cabinet" - section 6.5.

3 "Launch Plexus Administrator" - section 6.6.

4 "Start a new configuration" - section 6.7.

5 "Complete the peripheral card layout" - section 6.9.

6 "Configure Auto-Attendant Interface (AAI) card" - section 6.10.

7 "Configure Integrated Voice Processor (IVP) card" - section 6.11.

8 "Create users" - section 6.12 and "Create user groups" - section 6.13.

9 "Assign users to user groups" - section 6.14.

10 "Create trunks" - section 6.15 and "Create trunk groups" - section 6.16.

11 "Assign trunks to trunk groups" - section 6.17.

12 "Configure AEI extension ports" - section 6.18 and "Configure DEI extension ports" - section 6.19.

13 "Configure ATI trunk ports" - section 6.20.

14 "Configure system parameters" - section 6.21.

15 "Configure users" - section 6.22 and "Configure user groups" - section 6.23.

16 “Configure trunks” - section 6.24 and “Configure trunk groups” - section 6.25.

17 “Saving a configuration” - section 6.26.

18 “Establishing a link” - section 6.27.

19 “Uploading the configuration file” - section 6.28.

20 “Updating the system clock” - section 6.29.

21 “Updating Plexus key telephone programming” - section 6.30.

6.4 Determine the structure of the system

- Determine the flexible numbering plan that the phone system will follow. The numbering plan should be designed for optimum organization, clarity, and growth.
  - User IDs can consist of 2-5 digits as defined in the width parameter (Dial Plan tab).
  - The range is based upon the From and To parameters (Dial Plan tab). For example, if the Range is (From) 1 to (To) 4 with a Width of 3. IDs could be from 100 through 499. Likewise, if the Range is (From) 1 to (To) 4 with a Width of 4. IDs could be from 1000 through 4999.
  - The numbering plan will need to include: User; User Group, Trunk; Trunk Group; Auto Attendant; and Voice IDs.

Note
The Integrated Voice Processor (auto attendant and voice mailboxes) will only support IDs of 1-4 digits with the highest ID being 8999. If the auto attendant or voice mailbox features will be used, The user IDs must fall into this range.

- Determine the Access Digits for the system.
  - What number will the user dial to access an outside line (i.e., trunk)?
  - What number will the user dial to reach the system operator?
- Determine the number of groups on the system and their names.
  - What employees or stations will belong to what group?
  - User Groups = departments, organizational groups, teams, sections.
- Determine how many trunks will be available on the system.
  - Trunks = outside phone lines.
- Determine how the trunks should be grouped.
  - Trunk Groups = groups of outside phone lines.
- Determine if the organization will have an Auto-Attendant.
- Determine if the organization will have Plexus Integrated Voice Mail.
- Determine how calls will be routed to users and user groups.
- Determine how calls should be managed during the day, night, and weekends.
- Determine coverage (backup) for each user or user Group.

- Determine the features and privileges for each user (e.g., Hold Time-outs, Override capabilities, Call Intrusion capabilities, Toll Restrictions etc.).
- Determine how calls will be distributed to members of each user group.

6.5 Physically insert peripheral cards into the cabinet

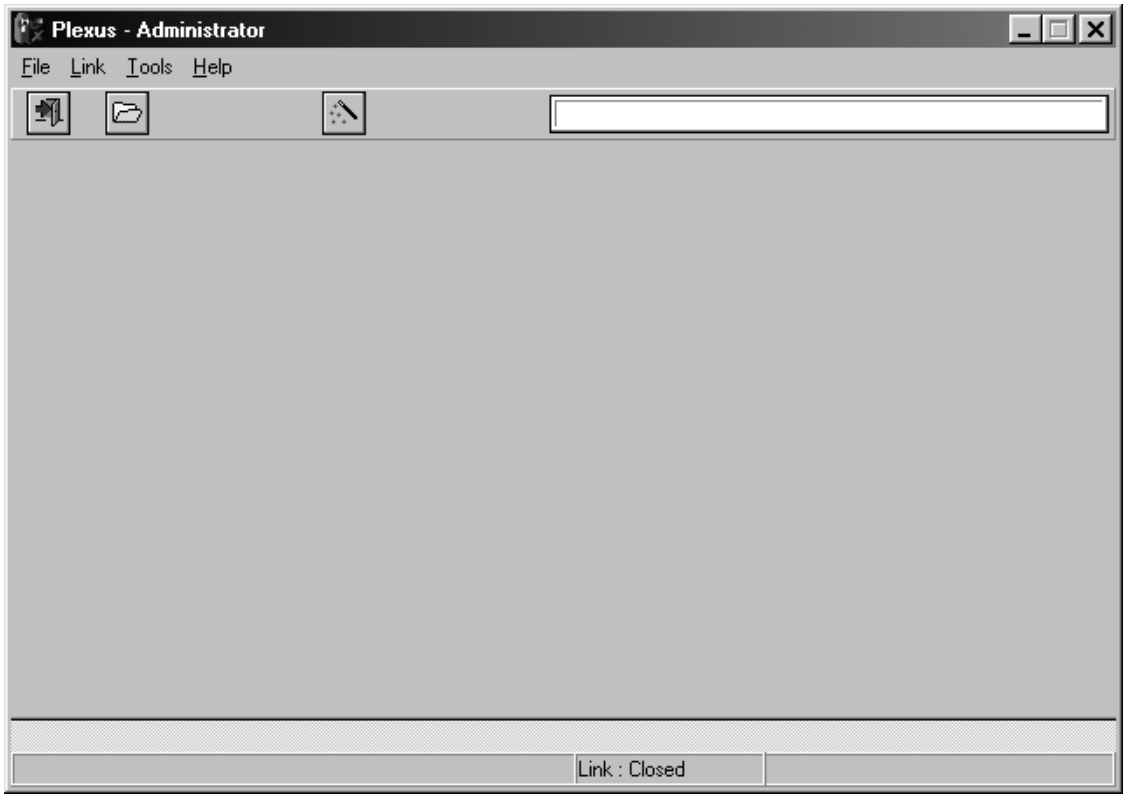
Refer to “Inserting Peripheral Cards” - section 4.3.

6.6 Launch Plexus Administrator

Launch the Plexus Administrator as follows:

1. Click on Start.
2. Select **Programs**.
3. Select Plexus.
4. Click on Plexus **Administrator**.

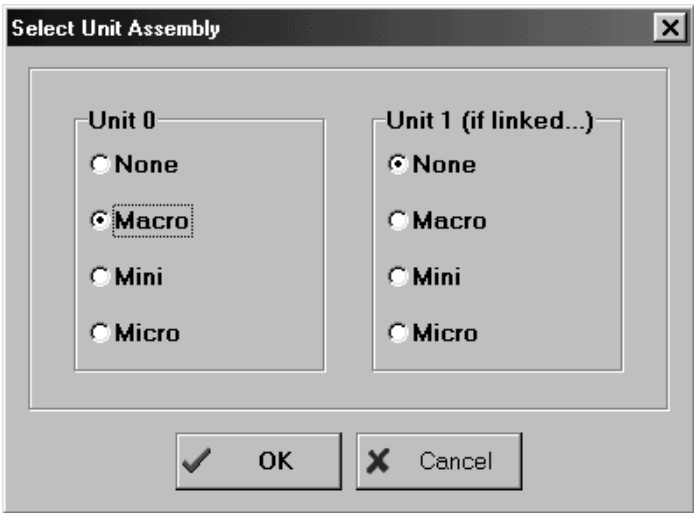
After launching Plexus Administrator, the following window displayed:



6.7 Start a new configuration

To start a new configuration:

1. From the **File** menu on the initial window, select **New**.
2. Select the type of System being configured (i.e., Macro or Micro).
3. If only one cabinet is in use, leave Unit 1 set to **None**.



6.8 Using an existing configuration file

If you are modifying a configuration using an existing configuration file:

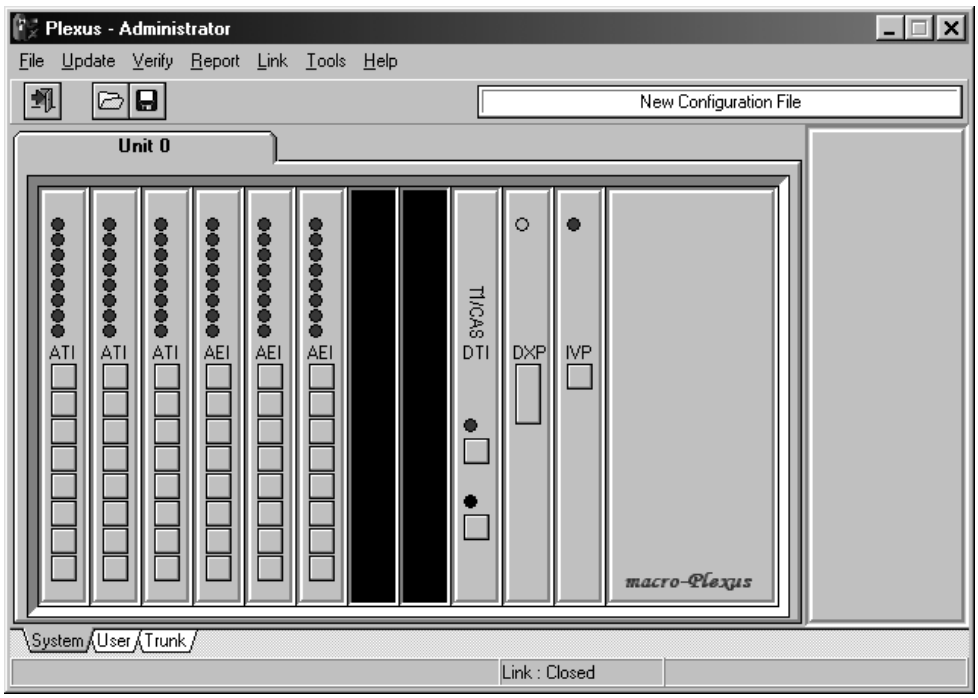
1. From the **File** menu on the initial window, select **Open**.
2. Select the desired configuration file.

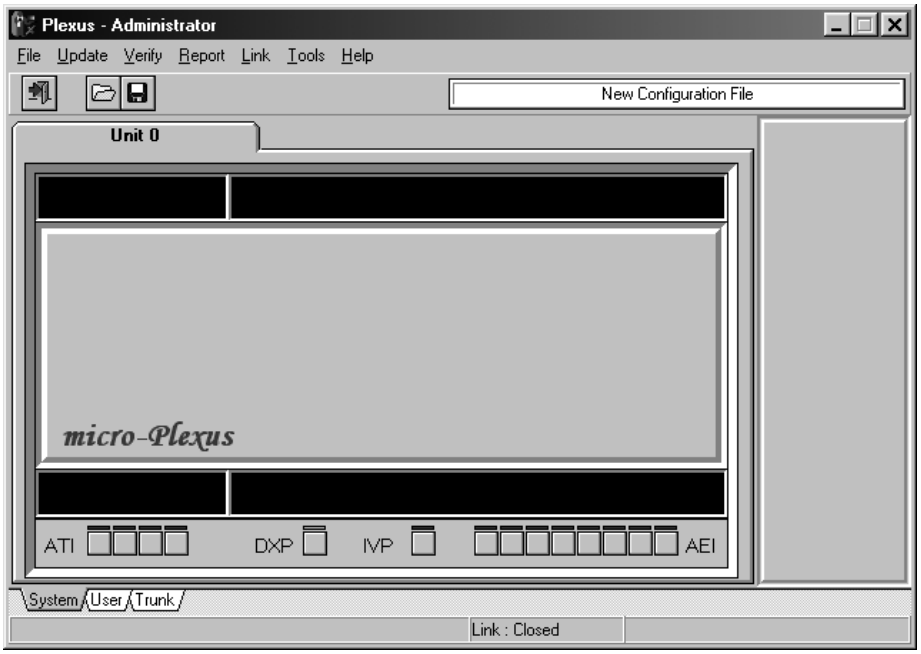
Plexus configuration files have an extension of **.zdb**. E.g., the Plexus Configuration Wizard will create a configuration file named **wiz-cfg.zdb**

6.9 Complete the peripheral card layout

6.9.1 Peripheral card layout

1 Click on the **System** tab to display the peripheral card layout.





2 With an image of the applicable cabinet (i.e. Macro Plexus or Micro Plexus) displayed, right-click on the various slots and select the card (ATI, AEI, etc.) that have been physically inserted.

An image of the selected card will appear in the slot.

3 Fill the slots on the Macro Plexus system from left to right as follows:

ATI-Analog Trunk Interface cards

AAI-Auto-Attendant Interface card (optional)

AEI-Analog Extension Interface cards

DEI-Digital Extension Interface cards

ILI-Inter-Unit Link Interface card (optional)-slot labeled 8

DXP-Digital Switch Processor-slot labeled “9”

IVP-Integrated Voice Processor card (optional)-slot labeled 10

For digital line Interface cards (T1, E1, ISDN) or Direct Inward Dial (DID) Trunk Interface cards, refer to the Interface card’s individual chapter.

4 For The Micro Plexus system, right-clicking in an empty slot will show the cards that are allowed for that slot. Other card choices will be grayed out.

Note
The Digital Switch Processor (DXP) card may only be placed in the slot in which it appears. The Inter-Unit Link Interface (ILI) card may only be placed in the rear slot of the Micro Plexus system and the slot labeled **8** on the Macro Plexus system.

Refer to the Installation section for details on inserting peripheral cards.

6.9.2 Alternative method to layout peripheral cards:

1 Physically insert peripheral cards.

2 Start Plexus Administrator.

3 From the **File** menu, select **New**.

4 Select the type of cabinet that is being configured, **Micro** or **Macro**.

5 From the **Link** menu, select **Open**.

6 From the **Verify** menu, select **Read**.

Upon completing the read, the following message will appear on the screen:

Verify: Read Operation Complete…

7 Click **OK**.

8 From the **Verify** menu, select **Synchronize**.

Note
The Digital Switch Processor (DXP) card may only be placed in the slot in which it appears. The Inter-Unit Link Interface (ILI) card may only be placed in the rear slot of the Micro Plexus system and the slot labeled **8** on the Macro Plexus system.

6.10 Configure Auto-Attendant Interface (AAI) card

Refer to chapter 15, "Automated Attendant Interface".

6.11 Configure Integrated Voice Processor (IVP) card

Refer to chapter 9, "Integrated Voice Processor".

6.12 Create users

6.12.1 Create users

1. Click on the **User** tab to display the User and User Group.
2. Click on the **Create** button in the User Window and enter a User ID sequence within the dial plan.
3. Create additional users by clicking on the **Create** button or by pressing **ENTER** on the keyboard.

6.12.2 About users

A user is an employee, employee alias, or device that will utilize the system for the purpose of placing or receiving calls.

- Jane Doe (employee)
- Salesperson X (employee alias)
- Device (**Dss** Console, modem, credit card scanner, fax machine)

Users can utilize several IDs by logging in or signing in to the system, depending on the functions they are to perform. Therefore there does **not** need to be a one-one relationship between user IDs and extension ports.

6.12.3 About the User ID sequence

The User ID sequence is the start number of the range of logical IDs to be assigned to system users. Users are assigned user IDs based on the established user ID sequence. The user ID sequence may be any number based on the Dial-Number Range (refer to System Parameters: Dial Plan). Default settings call for a range from 1 to 4 with a width of 3. Using these settings, the user ID sequence may be any number between 100 and 499.

Example
If the User ID sequence entered is 100, the first user created is assigned a user ID of 100. Each additional user created is assigned an ID based on the established sequence. Five users on a system with a user ID sequence of 100, would be 100, 101, 102, 103, and 104.

Tip
To insert or start a new sequence with a new starting number, click on the **Create** button while holding down **SHIFT** on the keyboard. Newly added sequences are automatically ordered.

See “Configure users” - section 6.22 for more details.

6.13 Create user groups

6.13.1 Create user groups

1. Click on the **User** tab to display the User and User Group dialog windows.
2. Click on the **Create** button in the User Group Window and enter a User Group ID sequence within the dial plan.
3. Create additional groups by clicking on the **Create** button or by pressing **ENTER** on the keyboard.

6.13.2 About user groups

- User Groups are groups of users (individuals or functions) who share certain system call routing features. User groups are typically used to establish certain departmental areas (e.g., Sales, Technical Support, etc.) for the purpose of receiving calls.

6.13.3 About the user group ID sequence

The user group ID sequence is the start number of the range of logical IDs to be assigned to user groups. The user group ID sequence may be any number based on the Dial-Number Range (refer to System Parameters: Dial Plan). Default settings call for a range from 1 to 4 with a width of 3. Using these settings, the user group ID sequence may be any number between 100 and 499.

If the user group ID sequence entered is 300, the first user group created is assigned a user group ID of 300. Each additional user group created is assigned an ID based on the established sequence. Five user groups on a system with a user group ID starting sequence of 300, would be 301, 302, 303 and 304.

Tip
To insert or start a new sequence with a new starting number, click on the **Create** button while holding down **SHIFT** on the keyboard. Newly added sequences are automatically ordered.

See “Configure user groups” - section 6.23 for more details.

6.14 Assign users to user groups

6.14.1 Assign users to user groups

Each user may be a member of any number of user groups.

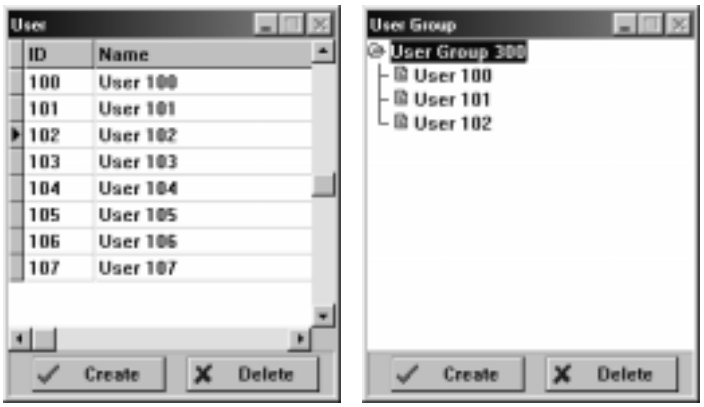
1. Ensure that users and user groups have been created.
2. Click on the user so that it becomes highlighted.
3. Click again on the user and hold down the left mouse button.
4. Drag the user over to the desired user group.

The user group name will become highlighted.

5. With the desired user group highlighted, release the mouse button.

6.14.2 About assigning users to user groups

The first user assigned to a user group converts the user group icon from a sheet of paper to a folder. Double-clicking on a user group folder icon displays the members of the user group.



Tip

To change the order in which users are listed in the user group, click on the user so that it becomes highlighted and drag it to the desired position in the user group.

CTRL and **SHIFT** may be used to select multiple users. Click on the first user so that it becomes highlighted. Hold down **CTRL** and select additional users in the desired order. Hold down **SHIFT** and use ↑ and ↓ to select a consecutive list of users. A selected group of users may be assigned to a user group in one drag and drop motion.

6.15 Create trunks

6.15.1 Create trunks

1. Click on the **Trunk** tab to display the Trunk and Trunk Group dialog window.
2. Click on the **Create** button in the trunk window and enter a trunk ID sequence within the dial plan.
3. Create additional trunks by clicking on the create button or by pressing **ENTER** on the keyboard.

6.15.2 About trunks

Trunks and trunk groups determine how a user will place external calls and determine how inbound calls are routed. Trunk groups are used for allocation purposes (e.g., to permit users access to groups of outside lines).

6.15.3 About the trunk ID sequence

The Trunk ID sequence is the start number of the range of logical IDs to be assigned to the trunks on the system. Trunks are assigned sequential Trunk IDs based on the established Trunk ID sequence. The trunk ID sequence may be any number based on the Dial-Number Range (refer to System Parameters: Dial Plan). Default settings call for a range from 1 to 4 with a width of 3. Using these settings, the trunk ID sequence may be any number between 100 and 499.

Example
If the Trunk ID sequence entered is 201, the first trunk created is assigned a Trunk ID of 201. Each additional trunk created is assigned an ID based on the established sequence. Five trunks on a system with a Trunk ID sequence of 201, would be 201, 202, 203, 204 and 205.

Tip
To insert or start a new sequence with a new starting number, click on the **Create** button while holding down **SHIFT** on the keyboard. Newly added sequences are automatically ordered.

See “Configure trunks” - section 6.24 for more details.

6.16 Create trunk groups

6.16.1 Create trunk groups

1. Click on the **Create** button in the trunk group Window and enter a trunk group ID sequence within the dial plan.
2. Create additional trunk groups by clicking on the **Create** button or by pressing **ENTER** on the keyboard.

6.16.2 About trunk groups

Trunks and trunk groups determine how a user will place external calls and determine how inbound calls are routed. Trunk groups are used for allocation purposes (e.g., to permit users access to groups of outside lines). Trunk groups will be assigned sequential trunk group IDs based on the established trunk group ID sequence.

6.16.3 About the trunk group ID sequence

The trunk group ID sequence is the start number of the range of logical IDs to be assigned to the trunk groups on the system. Trunk groups are assigned sequential trunk group IDs based on the established trunk group ID sequence. This sequence may be any number based on the Dial-Number Range (refer to System Parameters: Dial Plan). Default settings call for a range from 1 to 4 with a width of 3. Using these settings, the trunk group ID sequence may be any number between 100 and 499.

Example
If the trunk group ID sequence entered is 400, the first trunk group created is assigned a Trunk Group ID of 401. Each additional trunk group created is assigned an ID based on the established sequence. Five trunk groups on a system with a trunk group ID sequence of 401, would be 401, 402, 403, 404 and 405.

Tip
To insert or start a new sequence with a new starting number, click on the **Create** button while holding down **SHIFT** on the keyboard. Newly added sequences are automatically ordered.

See “Configure trunk groups” - section 6.25 for more details

6.17 Assign trunks to trunk groups

6.17.1 Assign trunks to trunk groups

To assign a trunk to a trunk group, proceed as follows. Each trunk may be a member of more than one trunk groups.

1. Ensure that both the trunk and trunk Group have been created.
2. Click on the trunk so that it becomes highlighted.
3. Click again on the trunk and hold down the left mouse button.
4. Drag the trunk over to the desired trunk group.

The trunk group name will become highlighted.

5. With the desired trunk group highlighted, release the mouse button.

6.17.2 About assigning trunks to trunk groups

The first trunk assigned to a trunk group converts the trunk group icon from a sheet of paper to a folder. Double-clicking on a trunk group folder icon displays the trunks in the trunk group.

Tip

Trunks should be listed in trunk groups in an order opposite to the hunt (rollover) order established by the phone company. Such an arrangement facilitates the use of the least busy trunks for outgoing calls. The order in which trunks are listed in the trunk group is changed by clicking on the trunk so that it becomes highlighted and dragging it to the desired position in the trunk group.

CTRL and **SHIFT** may be used to select multiple trunks. Click on the first trunk so that it becomes highlighted. Hold down **CTRL** and select additional trunks in the desired order. Hold down **SHIFT** and use ↑ and ↓ to select a consecutive list of trunks. A selected group of trunks may be assigned to a trunk group in one drag and drop motion.

6.18 Configure AEI extension ports

6.18.1 Configure extension ports

1 Click on the System tab.

An image of the system cabinet and the available peripheral cards will display.

2 Click on each of the buttons on the Analog Extension Interface (AEI) card to configure the extension ports.

3 Make appropriate entries and selections to each of the tabs.

Each LED image will appear red until the associated extension port has been assigned. Once assigned the LED image will appear green.

Tip

To alleviate the repetitive tasks associated with configuring multiple extension ports, right-click on a setting or field and select **Replicate**. Replicate automatically assigns a setting to all extension ports.

To alleviate the repetitive tasks associated with assigning Users to extension ports, right-click in the Port Assignment field on the General tab and select **Auto Assign**. Auto Assign automatically assigns Users to extension ports in a sequential fashion.

6.18.2 Extension port: General

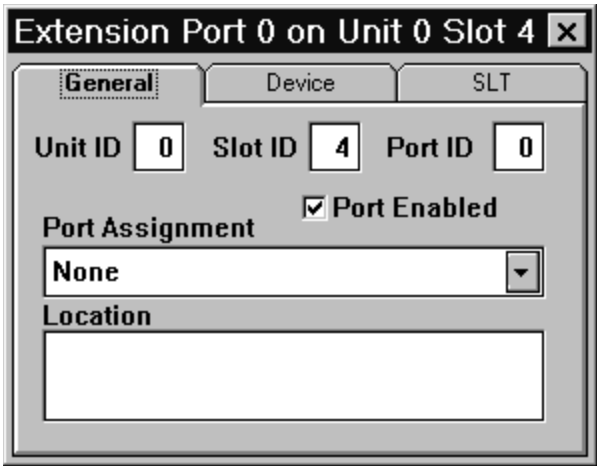


Table 6 - 1 Extension port: General

Unit ID	
Logical identifier for the unit (i.e., system cabinet) being configured. ***Note:*** Unless there is more than one system (i.e., a linked environment), this ID will be 0.	Automatically Assigned
Slot ID	
Logical identifier for the slot in which the Analog Extension Interface (AEI) card is inserted.	Automatically Assigned
Port ID	
Logical identifier for the extension port being configured.	Automatically Assigned
Port Enabled	
Enables the extension port.	Enabled*/Disable
Port Assignment	
The assigned user. Extension ports can be assigned to non-transient users to serve as their home station. **Tip**: Right mouse-click to utilize the Auto Assign feature.	None* User ID
Location	
Informal description or notes about the station location. E.g., Conference Room.	Alphanumeric

* = default setting

6.18.3 Extension port: Device

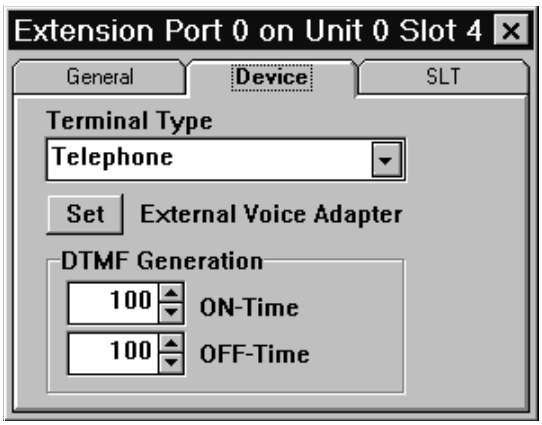


Table 6 - 2 Extension port: Device

Terminal Type		
The type of device connected to the extension port. ***Note***: if a modem, fax machine, credit card scanner or other device is to be connected, select Telephone.	Telephone* Door-Phone External Voice-Adapter	
Set External Voice Adapter		
If an external voice-adapter is selected above, click on this button to configure the adapter. ***Note:*** External voice-adapter refers to a third-party voice mail system.		
DTMF Generation		
The ON/OFF time for DTMF digits sent to the external voice adapter.	0-255 ms	100 ms*

* = default settings

6.18.3.1 Set external voice adapter

1. Click on the image of the extension port.
2. Click on the **Device** tab.
3. Click on the **Set** button to display the Voice Adapter dialog window.
4. Configure the external voice adapter by making appropriate entries and selections on each of the voice adapter tabs.

6.18.4 Extension port: Voice Adapter, Access

The Access tab addresses DTMF digits sent to the external voice adapter when system entities attempt to access the extension port either as direct internal calls or redirected calls.

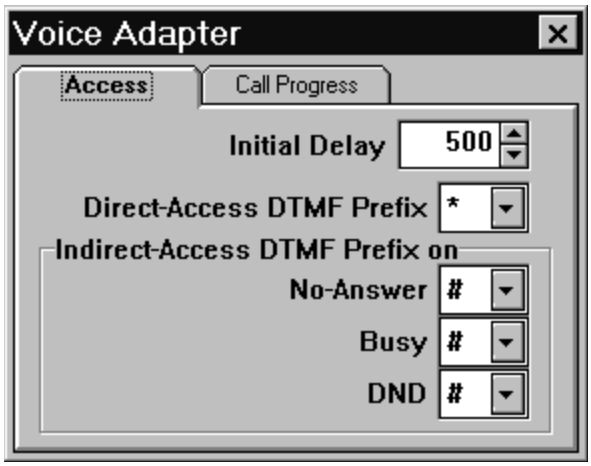


Table 6 - 3 Extension port: Voice Adapter, Access

Initial Delay		
The duration of the initial delay when sending DTMF to the external voice adapter.	50-9000 ms	500*
Direct – Access DTMF Prefix		
The DTMF digit sent to the external voice adapter when a user or trunk directly accesses the extension port.	0-9, A-D, * (default), X, #	
Indirect – Access DTMF Prefix On		
The DTMF digit sent to the external voice adapter when a user or trunk indirectly accesses the extension port (through a user's coverage).	0-9, A-D, *, X, #*	

* = default settings

6.18.5 Extension port: Voice Adapter, Call Progress

The Call Progress tab enables the sending of DTMF digits to the external voice adapter to represent call progress tones detected when the associated extension port attempts to access other system entities (e.g., users, trunks).

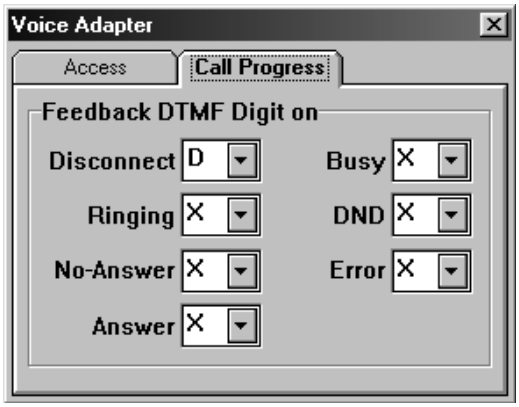


Table 6 - 4 Extension port: Voice Adapter, Call Progress

Feedback DTMF Digit on		
The DTMF digit which represents the associated call progress tone. ***Note:*** X = send extension number	Disconnect Ringing No-Answer Answer Busy DND Error	D* X* X* X* X* X* X*

* = default setting

6.18.6 Extension port: SLT

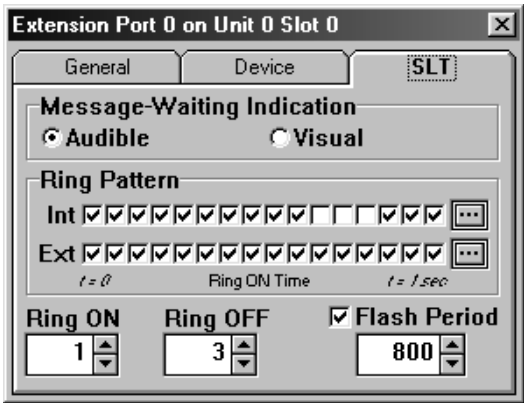


Table 6 - 5 Extension port: SLT

Message-Waiting Indication	
The method of message waiting indication utilized on the (SLT) Single Line Telephone. **Note:** To utilize the visual method, the SLT must have a message waiting lamp compatible with the Plexus signaling format.	Audible* Visual
Ring Pattern Internal	
The ring pattern for the ring **On** position for internal calls. ***Note:*** Plexus Key Telephone users have the option of changing ring pattern directly from their phones. Choose **Int. Ring Type**.	60% on, 20% off, 20% on* or Continuous ON
Ring Pattern External	
The ring pattern for external calls. ***Note:*** Users of Key Telephones have the option of changing ring pattern directly from their phones. Choose **Ext. Ring Type**.	60% on, 20% off, 20% on or Continuous ON
Ring ON	
Period during which the ring tone will be on.	1-2 seconds 1 second *

Table 6 - 5 Extension port: SLT (continued)

Flash Period		
Allows SLTs to do an internal flash	enabled*/ disabled	
Flash Period		
The duration of an internal flash signal. ***Note***: Applies to SLTs	50-10000 ms	800 ms *

* = default setting

6.20 Configure ATI trunk ports

6.20.1 Configure ATI trunk ports

1 Click on the **System** tab.

An image of the system cabinet and the available peripheral cards will be displayed.

2 Click on each of the buttons on the Analog Trunk Interface (ATI) card to configure the trunk ports.

3 Make appropriate entries and selections on each of the tabs.

Tip
To alleviate the repetitive tasks associated with configuring multiple trunk ports, right-click on a setting or field and select **Replicate**. Replicate automatically assigns a setting to all trunk ports.

To alleviate the repetitive tasks associated with assigning trunks to trunk ports, right-click in the Port Assignment field on the General tab and select **Auto Assign**. Auto Assign automatically assigns trunks to trunk ports in a sequential fashion.

Note
Each LED image will appear red until the associated trunk port has been assigned. Once assigned the LED image will appear green.

6.20.2 Trunk ports: General

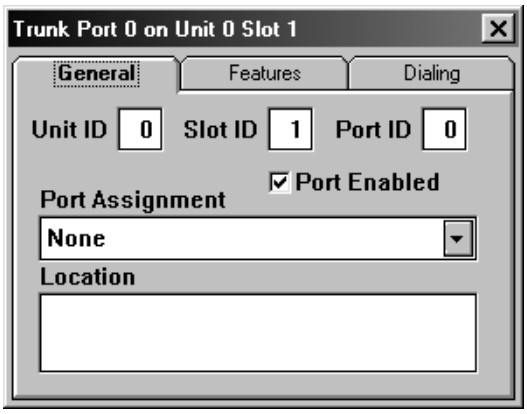


Table 6 - 7 Trunk ports: General

Unit ID	
Logical identifier for the unit (i.e., system cabinet) being configured. ***Note:*** Unless there is more than one system (i.e., a linked environment), this ID will be 0.	Automatically Assigned
Slot ID	
Logical identifier for the slot in which the Analog Trunk Interface (ATI) card is inserted.	Automatically Assigned
Port ID	
Logical identifier for the trunk port being configured.	Automatically Assigned
Port Enabled	
Enables the trunk port.	Enable */Disable
Port Assignment	
The assigned trunk. **Tip**: Right mouse-click to utilize the Auto Assign feature.	None * Trunk ID
Location	
Informal description or notes about the location of the trunk wiring. e.g., Phone closet	Alphanumeric

* = default settings

6.20.3 Trunk ports: Features

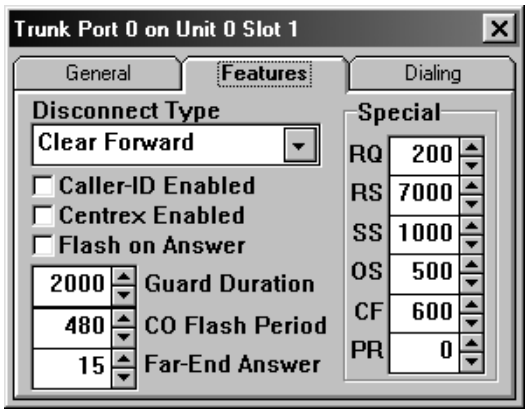


Table 6 - 8 Trunk ports: Features

Disconnect Type	
The type of disconnect supervision utilized by the local telephone company. ***Note:*** Check with the local telephone company to determine which type of disconnect supervision is used on the lines. All trunk ports should be set to the same type of disconnect supervision. In the U.S., the primary disconnect is Clear Forward.	None Clear Forward* Polarity Reversal Disconnect Tone
Caller-ID Enabled	
When enabled, the system will detect the Caller ID signal for each incoming call on the trunk. ***Note:*** Caller ID can only be enabled on Caller ID equipped Analog Trunk Interface (ATI) cards. Caller ID service must be activated on the line in order for a Caller ID signal to be present.	Enable/Disable*
Centrex Enabled	
Identifies if the trunk has Centrex capabilities with the phone company. When enabled, the system will allow the sending of a 'flash' signal for externally forwarded or transferred calls.	Enable/Disable *

Table 6 - 8 Trunk ports: Features (continued)

Flash on Answer		
Used to 'flash' the CO on seizing an inbound call. This is used to signal the CO to reject certain types of calls (e.g., collect calls).	Enable/ disable*	
Guard Duration		
The period of time that a trunk is unavailable following the completion of a call.	100-2000 ms	2000 ms *
CO Flash Period		
The duration of the 'flash' signal as sent to the central office. ***Note:*** Check with the local telephone company to determine the proper duration of the 'flash' signal.	50-10000 ms	480 ms *
Far-End Answer		
When the system starts timing external calls for SMDR output information if the system has not already received indication that the call has been answered by the remote party.	0-255 seconds	15 seconds*
RQ (Ring Qualifier)		
The duration necessary for voltage on the line to be considered a ring tone by the system. Voltage of a shorter duration is considered line noise.	10-1000 ms	200 ms *
RS (Ring Silence)		
The duration of the of the interval between ring tones used by the system to determine that a call was terminated.	500-15000 ms	7000 ms *
SS (Seize Settling)		
The amount of time the system must wait once a trunk port has been opened (e.g., seizing a trunk) to reach the nominal operating condition. During this period, disconnect signals are ignored, and digit buffering is in effect.	100-2000 ms	1000 ms *
OS (Central Office Setup)		

Table 6 - 8 Trunk ports: Features (continued)

The amount of additional time following Seize Settling (SS) that the system must wait to allow the Central Office to reach the nominal operating condition. During this period, disconnect signals are ignored and digit buffering is in effect. Upon expiration of the OS time-out, the buffer is emptied and digits are sent to the Central Office.	0-8000 ms	500 ms *
CF (Clear Forward)		
The duration of a break in signal from the central office that represents a terminated call.	0-2000 ms	600 ms *
PR (Polarity Reversal)		
Disqualification period used in detecting polarity reversals on the line. The PR period begins as soon as a polarity reversal is detected. If a second reversal is detected within the period, the initial PR is ignored or disqualified.	0-3000 ms	0 ms *

* = default settings

6.20.4 Trunk port: Dialing

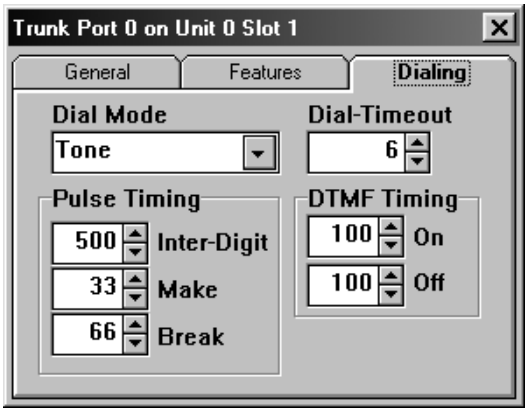


Table 6 - 9 Trunk port: Dialing

Dial Mode		
The type of dialing utilized by the trunk port. ***Note:*** Check with the local telephone company to determine the dial mode(s) supported on the line. If the line supports both Tone and Pulse, the port should be set to Tone.	Tone * Pulse	
Dial-Timeout		
The period of time that must elapse before the system assumes that the dialing sequence is complete. ***Note:*** Buffering and pulse-to-tone or tone-to-pulse conversion terminates according to the network dial timeout.	0-255 seconds	6 seconds *
Pulse Timing		
Timing parameters associated with components of a pulse string.		
Inter-Digit	50-1000	500
Make	10-255	33
Break	10-255 ms	66 ms *
DTMF Timing		
Timing parameters associated with the various components of a DTMF tone.		
On	50-255	100
Off	50-255 ms	100 ms *

* = default settings

6.21 Configure system parameters

6.21.1 Configure system parameters

1 Click on the **System** tab.

An image of the system cabinet and the available peripheral cards will be displayed.

2 Click on the button on the Digital Switch Processor (DXP) card.

3 Make appropriate entries and selections on each of the tabs.

6.21.2 Posting changes to system parameters

The following five buttons appear at the bottom of some system parameters tabs to prevent configuration errors from being posted to the system. The **Insert Record** button must be clicked on prior to adding any recording and the **Insert** or **Delete Record** button must be clicked on prior to making any changes.

Table 6 - 10 System Parameters: buttons

Edit record	
	Enables a record to be edited.
Post edit	
	Posts the changes made to the record.
Cancel edit	
	Cancels the changes made to the record.
Insert record	
	Inserts a new blank record in a list.
Delete record	
	Deletes a record from a list.

6.21.3 System parameters: General, Passwords

Table 6 - 11 System parameters: General, Passwords

Admin Password		
Password to Access system programming	Numeric, up to 12 digits	123456*
DISA Password		
Password to access Direct Inward System Access and Remote Station Service.	Numeric, up to 12 digits	123456*

* = default settings

6.21.4 System parameters: General, CTi Settings

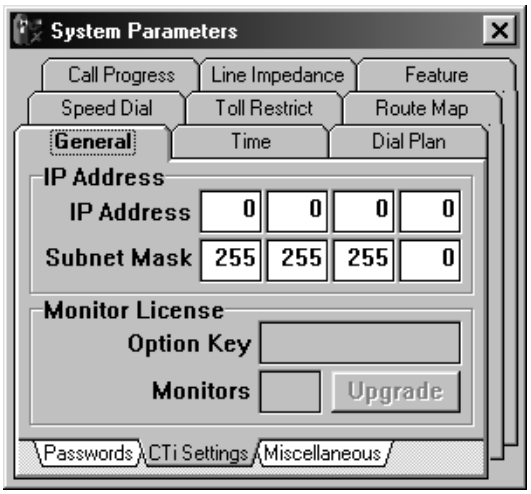


Table 6 - 12 System parameters: General, CTi Settings

IP Address		
System IP Address for CTi Interface	Numeric, 1-3 digits per field	0, 0, 0, 0*
Subnet Mask		
Subnet Mask for CTi Interface	Numeric, 1-3 digits per field	255, 255, 255, 0*
Option Key		
Unique software key for CTi Interface	Alpha-Numeric	None*
Monitors		
Number of system resources that can be monitored with current CTi software	Numeric	

* = default settings

Note
See chapter 16, “CTi Interface” for information on programming CTi Settings

6.21.5 System Parameters: General, Miscellaneous

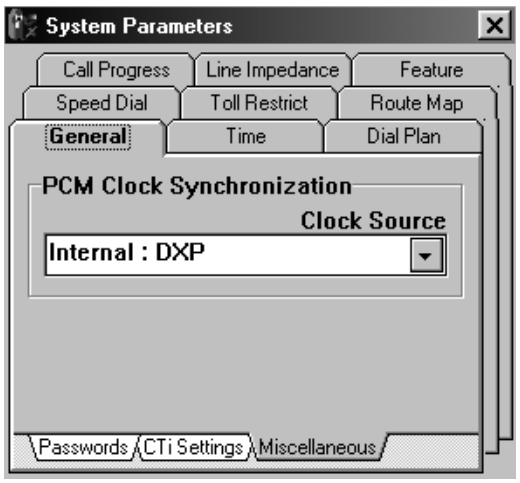


Table 6 - 13 System Parameters: General, Miscellaneous

Clock Source	
Clock synchronization for digital trunk cards	varies with types of digital trunk cards in the system

Note
See digital trunk card chapters for information on programming this parameter.

6.21.6 System parameters: Time, Mode

To designate call routing for each trunk based upon system time, see “Trunk: Routing: Voice” - section 6.24.2 and “Trunk: Routing: Fax” - section 6.24.3.

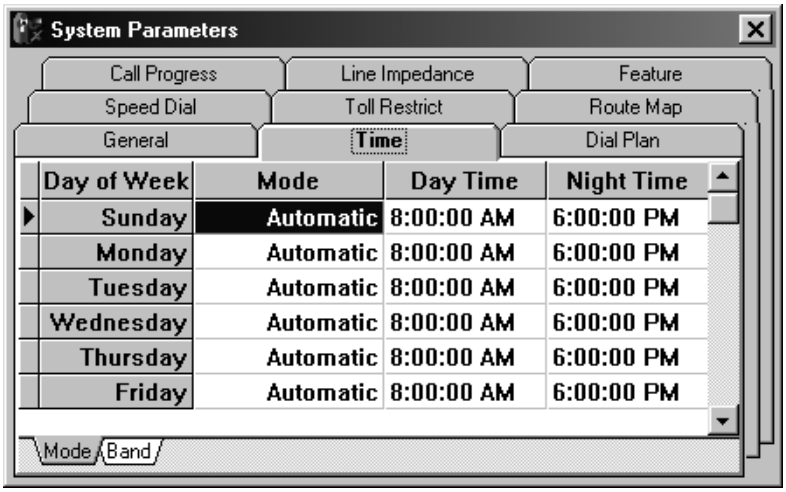


Table 6 - 14 System parameters: Time, Mode

Mode		
Sets whether the time automatically switches modes or if the system will stay in day mode or night mode for the entire day.	Automatic*, Day, Night	
Day Time		
Time that the system switches to day mode if Automatic switching is enabled for that day.	Time; Hour, Minute, AM/PM	8:00 AM*
Night Time		
Time that the system switches to night mode if Automatic switching is enabled for that day.	Time; Hour, Minute, AM/PM	6:00 PM*

* = default settings

6.21.7 System Parameters: Time, Band

Time, Band is user to designate outbound calls based upon least cost routing, see chapter 14, "Least Cost Routing".

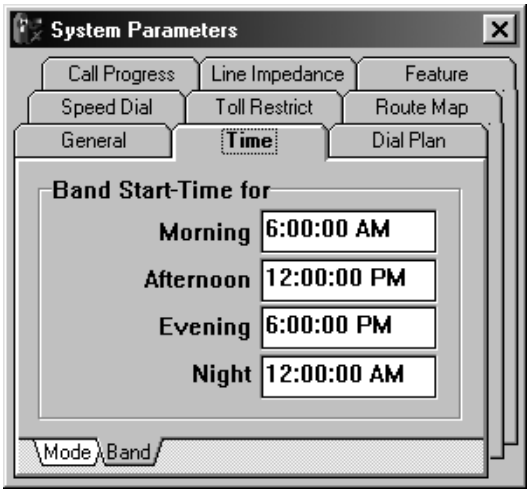


Table 6 - 15 System Parameters: Time, Band

Morning		
Time that Least Cost Routing starts following the Morning table to route calls.	Time; Hour, Minute, Seconds, AM/PM	6:00:00 AM*
Afternoon		
Time that Least Cost Routing starts following the Afternoon table to route calls.	Time; Hour, Minute, Seconds, AM/PM	12:00:00 PM*
Evening		
Time that Least Cost Routing starts following the Evening table to route calls.	Time; Hour, Minute, Seconds, AM/PM	6:00:00 PM*
Night		
Time that Least Cost Routing starts following the Night table to route calls.	Time; Hour, Minute, Seconds, AM/PM	12:00:00 AM*

* = default settings

6.21.8 System parameters: Dial Plan

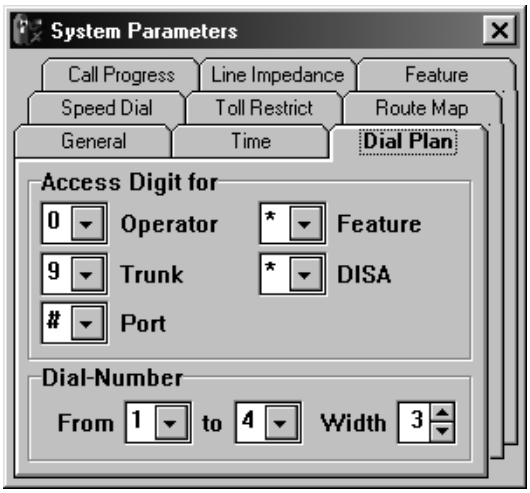


Table 6 - 16 System parameters: Dial Plan

Access Digit for		
Establishes the single digit dialed to…		
Reach the Operator	0-9, *, #	0 *
Access a outside line (Trunk)	0-9, *, #	9 *
Precede a Feature code	0-9, *, #	* (default)
Access a Port	0-9, *, #	# d
Utilize DISA	0-9, *, #	* (default)
Note: A unique digit must be assigned to each of the five access options, with the exception of Utilize DISA.		
Dial-Number Range		
The flexible numbering plan parameters used for assigning logical IDs to system resources such as User IDs, User Group IDs, Trunk IDs, Trunk Group IDs, Voice IDs, and Auto -Attendant IDs.		
Range From	0-9	1 *
Range To	0-9	4 *
Width	1-5	3 *
Note: The dial plan is uniform. For example, if there is a **Range From** 1, and a **Range to** 3, with a width of 4, the possible dial ranges are from 1000 to 3999. Values such as 100 or 10 are not valid.		

* = default settings

6.21.9 System parameters: Speed Dial

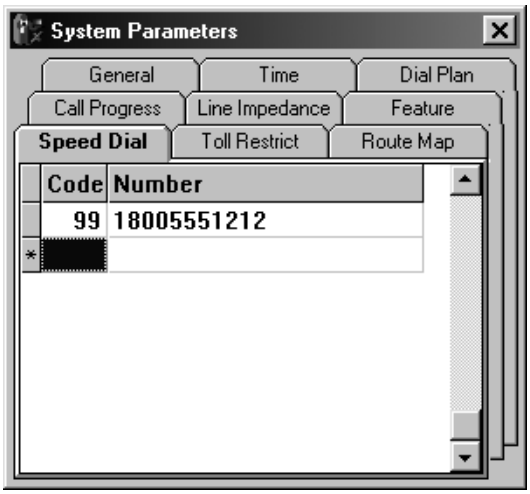


Table 6 - 17 System parameters: Speed Dial

Code	
Digits used to execute each system speed dial number. ***Note***: Single digits (1-9) are actually mapped with a preceding zero. For example, when utilizing code **1**, the user dials **01**. ***Note:*** Plexus Key Telephone users may program additional personal speed dial numbers using the speed dial memory on the key telephone [Refer to “User: Features, Settings” - section 6.22.3].	0-99
Number	
The telephone number that will be dialed when the speed dial code is entered. **Note:** The Plexus system has the capacity to store approximately 100 20-digit speed dial numbers. If the speed dial numbers are generally more than 20 digits, fewer numbers may be stored. Commas (i.e., pauses) count as digits.	Phone #

6.21.10 System parameters: Toll Restrict, Deny Dialing

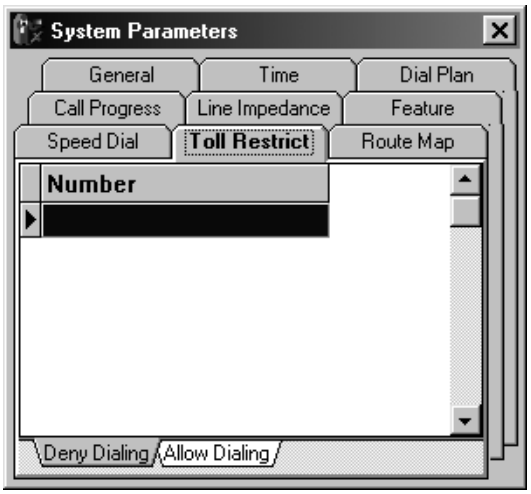


Table 6 - 18 System parameters: Toll Restrict, Deny Dialing

Number	
List of generally denied phone numbers and prefixes applicable to unassigned extensions and system out dialing. E.g., 1-900 would terminate calls with this prefix unless the specific number dialed is allowed in the system allow dialing table. ***Note***: System-dialed calls as in the case of Remote Call Forwarding, and Remote Telephone Notification, and Pager Notification are subject to the system deny and allow dialing tables	Phone # / Phone # prefix

6.21.11 System parameters: Toll Restrict, Allow Dialing

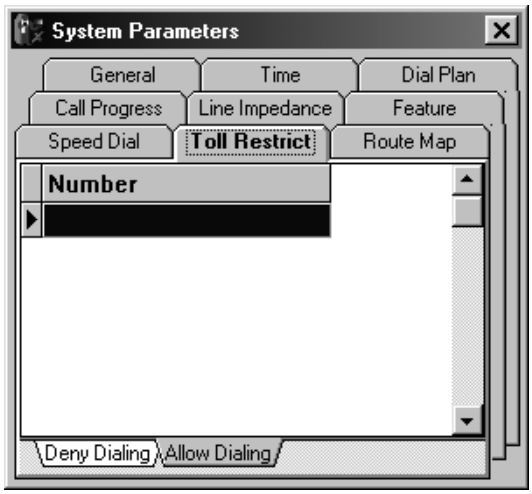
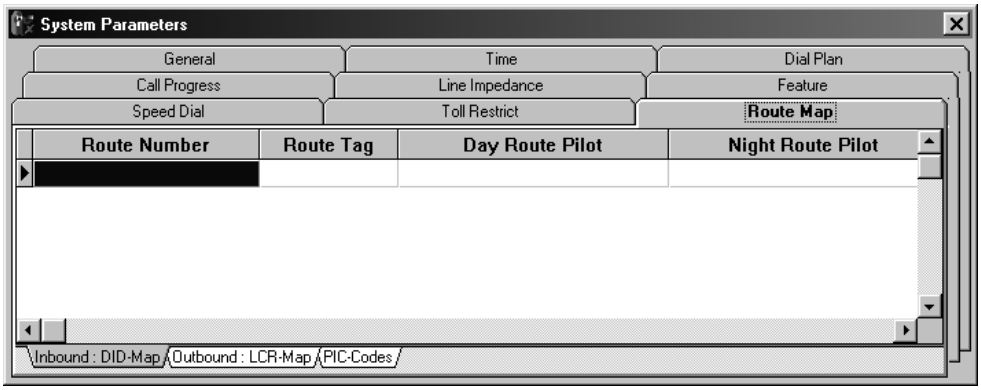


Table 6 - 19 System parameters: Toll Restrict, Allow Dialing

Number	
List of specifically allowed phone numbers applicable to unassigned extensions and system out dialing. These numbers will override the Deny Dialing table.	Phone # / Phone # prefix

6.21.12 System Parameters: Route Map, Inbound: DID-Map



The Inbound: DID-Map is used to route calls for Caller ID / ANI Routing, Mapped DID / DNIS Routing, and DID Routing. See chapter 13, "Direct Inward Dialing".

Table 6 - 20 System Parameters: Route Map, Inbound: DID-Map

Route Number	
The Caller ID number, DID number, DNIS number, or ANI number sent by the incoming call.	numeric, 1-20 digits
Route Tag	
Identifying name or information for this route number	alphanumeric, 0- 10 digits
Day Route Pilot	
System resource that the call will be routed to during the day mode.	None * User ID User Group ID AutoAttendant ID Voice ID
Night Route Pilot	
System resource that the call will be routed to during the night mode.	None * User ID User Group ID AutoAttendant ID Voice ID

* = default settings

6.21.13 System Parameters: Route Map, Outbound: LCR-Map

Note
Least Cost Routing is covered in detail in chapter 14, "Least Cost Routing". Refer to this chapter before configuring these settings.

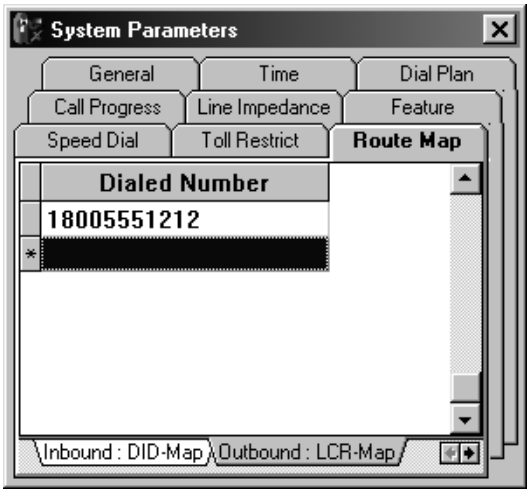


Table 6 - 21 System Parameters: Route Map, Outbound: LCR-Map

Dialed Number	
Number or prefix of the number dialed by a user with Least Cost Routing enabled.	Numeric, None*

* = default setting

Double clicking in a completed Dialed Number field brings up the LCR-Map Definition window.

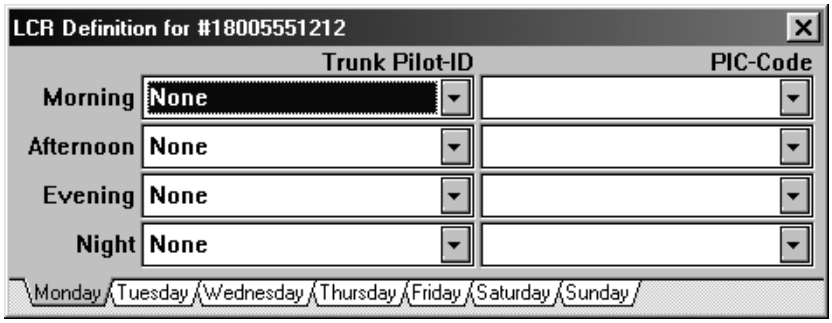


Table 6 - 22 System Parameters: Route Map, Outbound: LCR-Map: LCR Definitions

Trunk Pilot-ID	
Trunk or trunk group that will be accessed for this time band when the user dials the defined number.	None* Trunk ID Trunk Group ID
PIC-Code	
PIC Code that will be dialed with the number dialed by the user when the trunk is accessed.	Numeric, None*

* = default settings

6.21.14 System Parameters: Route Map, PIC-Codes

Note
Least Cost Routing is covered in detail in chapter 14, "Least Cost Routing". Refer to this chapter before configuring these settings.

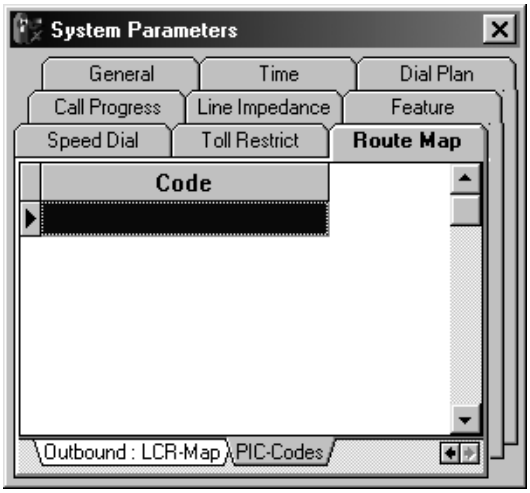


Table 6 - 23 System Parameters: Route Map, PIC-Codes

Code	
Available PIC Codes that will be dialed with the number dialed by the user when the trunk is accessed with Least Cost Routing.	Numeric, None*

* = default settings

6.21.15 System parameters: Call Progress, Filter

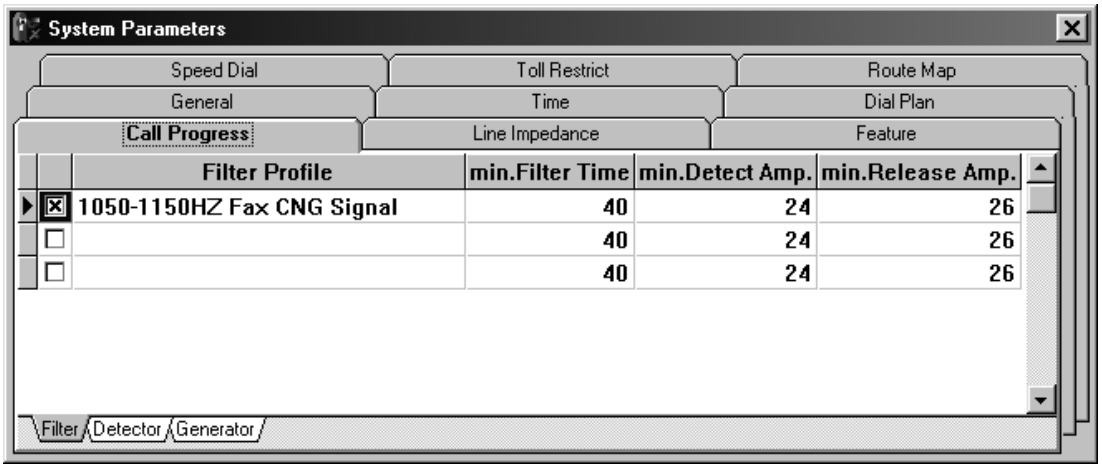


Table 6 - 24 System parameters: Call Progress, Filter

Check Box		
Makes the corresponding filter available or unavailable.	Available * Unavailable	
Filter Profile		
Description or name of the frequency range filter profile. ***Note***: Filters are used to detect such external tones as busy, dial, ring back, disconnect and CNG.		
min. Filter Time		
Period during which the system will attempt to detect the frequency range defined in the filter.	0-500	200 ms *
min. Detect Amp		
Minimum amplitude level required for the system to detect the defined frequency range.	0-40	24dB *
min. Release Amp.		
Minimum amplitude level at which the system will terminate detection of the defined frequency range. Note: It may be necessary to scroll right to see all of the parameters.	0-40	26dB *

* = default settings

6.21.16 System parameters: Call Progress, Detector

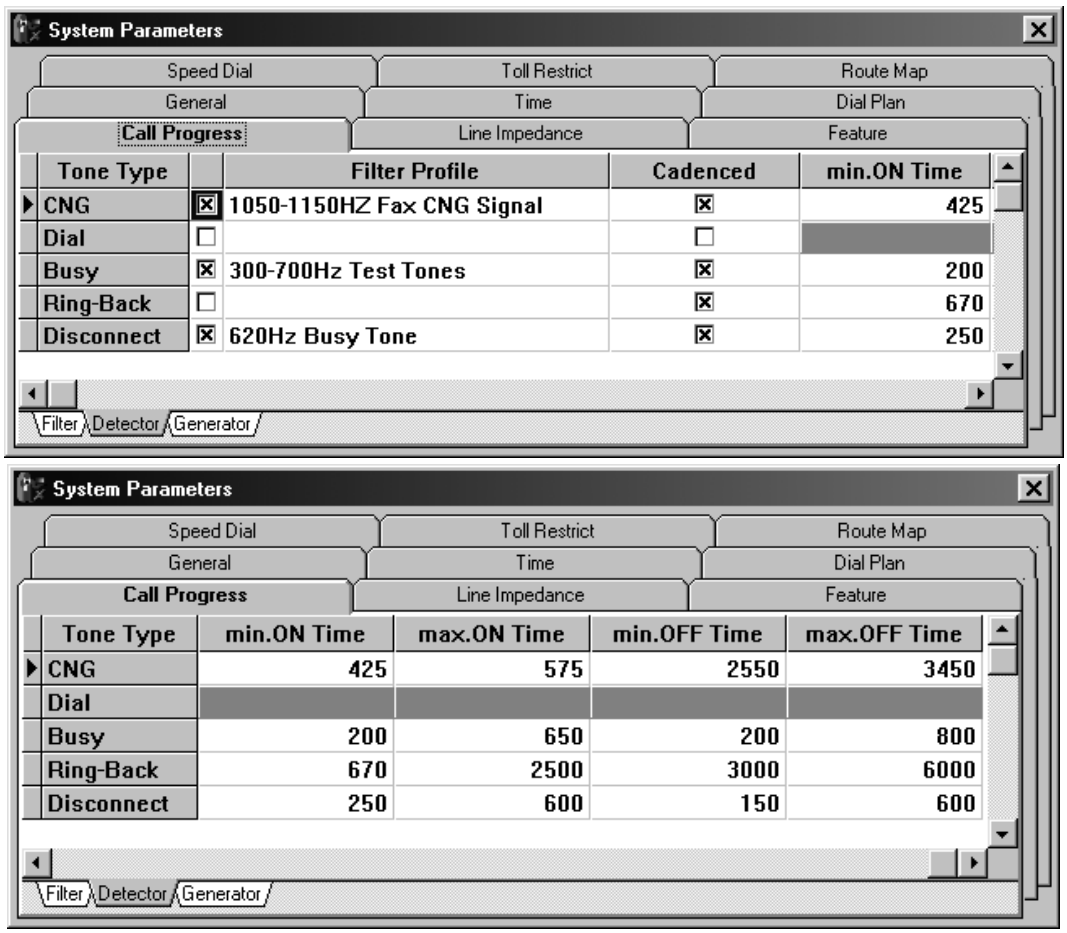


Table 6 - 25 System parameters: Call Progress, Detector

Tone Type	
Type of tone detected with the applicable filter.	
Check Box	
Activates or deactivates the filter profile for the applicable tone.	Active/Inactive
Filter Profile	
Description or name of the frequency range filter profile.	

Table 6 - 25 System parameters: Call Progress, Detector

Cadenced	
Enables the system to detect either a cadenced or continuous tone. ***Note***: If cadenced is enabled the following five parameters must be addressed in defining the minimum number of cycles and the ON and OFF positions of the cadence.	Enable/Disable
min. ON Time	
Minimum range for the ON position of a cadenced tone. See Chart B: min ON Time System Defaults and Ranges for details.	
max ON Time	
Maximum range for the ON position of a cadenced tone. See Chart C: max ON Time System Defaults and Ranges for details.	
min. OFF Time.	
Minimum range for the OFF position of a cadenced tone. See Chart D: min OFF Time System Defaults and Ranges for details.	
max OFF Time	
Maximum range for the OFF position of a cadenced tone. See Chart E: max OFF Time System Defaults and Ranges for details. Note: It may be necessary to scroll right to see all of the parameters.	

6.21.16.1 Chart A: System defaults and ranges (in milliseconds)

Table 6 - 26 System parameters: Call Progress, Detector

min ON Times	**CNG**	
	425 *	0-20000 (increments of 10)
	Dial	
	0 *	0-20000 (increments of 10)
	Busy	
	200 *	0-20000 (increments of 10)
	Ring Back	
	670 *	0-20000 (increments of 10)
	Disconnect	
	200 *	0-20000 (increments of 10)

* = default settings

6.21.16.2 Chart B: System defaults and ranges (in milliseconds)

Table 6 - 27 System parameters: Call Progress, Detector

min ON Time	**CNG**	
	575 *	0-20000 (increments of 10)
	Dial	
	0 *	0-20000 (increments of 10)
	Busy	
	650 *	0-20000 (increments of 10)
	Ring Back	
	2500 *	0-20000 (increments of 10)
	Disconnect	
	650 *	0-20000 (increments of 10)

* = default settings

6.21.16.3 Chart C: System defaults and ranges (in milliseconds)

Table 6 - 28 System parameters: Call Progress, Detector

Min OFF Time	**CNG**	
	2550 *	0-20000 (increments of 10)
	Dial	
	0 *	0-20000 (increments of 10)
	Busy	
	200 *	0-20000 (increments of 10)
	Ring Back	
	3000 *	0-20000 (increments of 10)
	Disconnect	
	200 *	0-20000 (increments of 10)

* = default settings

6.21.16.4 Chart D: System defaults and ranges (in milliseconds)

Table 6 - 29 System parameters: Call Progress, Detector

max OFF Time	**CNG**	
	3450 *	**0-20000 (increments of 10)**
	Dial	
	0 *	0-20000 (increments of 10)
	Busy	
	800 *	0-20000 (increments of 10)
	Ring Back	
	6000 *	0-20000 (increments of 10)
	Disconnect	
	800 *	0-20000 (increments of 10)

* = default settings

6.21.17 System parameters: Call Progress, Generator, Ringback

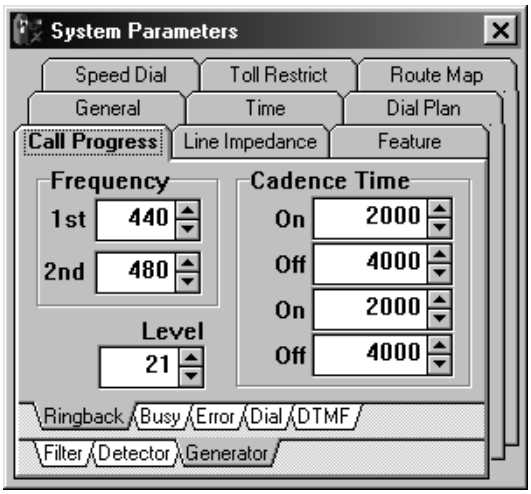


Table 6 - 30 System parameters: Call Progress, Generator, Ringback

Ringback Frequency		
Frequency settings for the internal ringback tones.	0-4000 Hz	1st 440 Hz * 2nd 480 Hz *
Ringback Cadenced Time		
Period for the ON and OFF positions of a cadenced ringback tone. ***Note***: Control is given for two cycles to allow for two different ON or OFF periods.	0-4095 ms	On 2000 ms * Off 4000 ms *
Ringback Level		
Amplitude level for internal ringback tones.	0-70 dB	21 dB *

* = default settings

6.21.18 System parameters: Call Progress, Generator, Busy

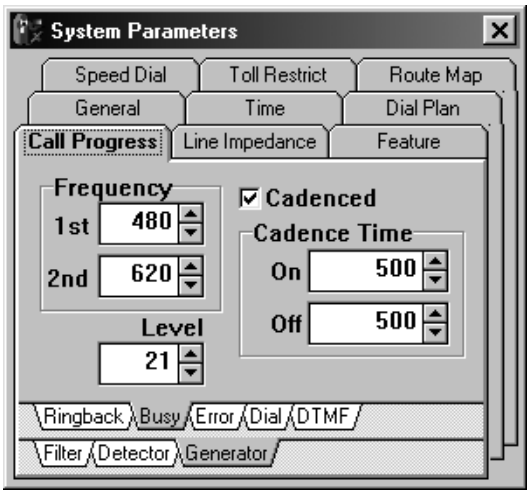


Table 6 - 31 System parameters: Call Progress, Generator, Busy

Busy Frequency		
Frequency settings for internal busy tone.	0-4000 Hz	1st 480 Hz * 2nd 620 Hz *
Busy Cadenced		
Enables/disables a cadenced busy tone.	Enable/Disable *	
Busy Cadence Time		
Period for the ON and OFF positions of a cadenced busy tone.	0-4095 ms	On 500 ms * Off 500 ms *
Busy Level		
Amplitude level for internal busy tones.	0-70 dB	21 dB *

* = default settings

6.21.19 System parameters: Call Progress, Generator, Error

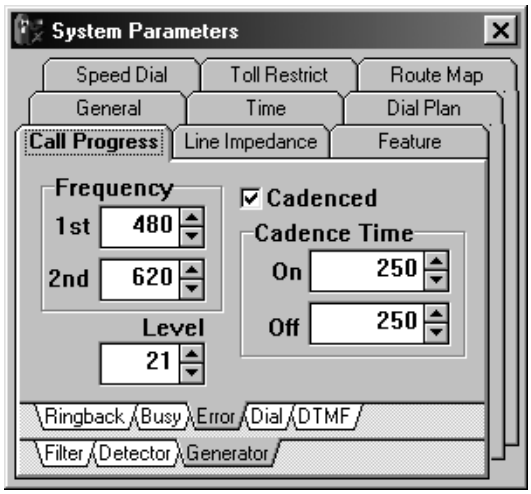


Table 6 - 32 System parameters: Call Progress, Generator, Error

Error Frequency		
Frequency setting for internal error tone.	0-4000 Hz	1st 480 Hz * 2nd 620 Hz *
Error Cadenced		
Enables/disables a cadenced error.	Enabled /Disable	
Error Cadence Time		
Period for the ON and OFF positions of a cadenced error tone.	0-4095 ms	On 250 ms * Off 250 ms *
Error Level		
Amplitude level internal error tone.	0-70 dB	21dB *

* = default settings

6.21.20 System parameters: Call Progress, Generator, Dial

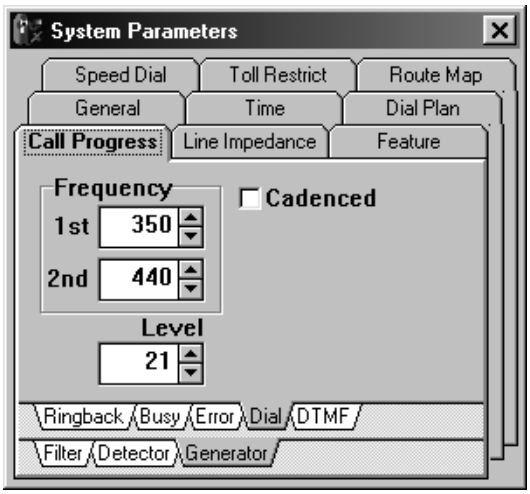


Table 6 - 33 System parameters: Call Progress, Generator, Dial

Dial Frequency		
Frequency settings for internal dial tone.	0-4000 Hz	1st 350 Hz * 2nd 440 Hz *
Dial Cadenced		
Enables/disables a cadenced dial tone.	Enable/Disable *	
Dial Cadence Time		
Period for the ON and OFF positions of a cadenced dial tone.	0-4095 ms	On 0 ms * Off 0 ms *
Dial Level		
Amplitude level for internal dial tone.	0-70 (-dB)	21 (-dB) *

* = default settings

6.21.21 System parameters: Call Progress, Generator, DTMF

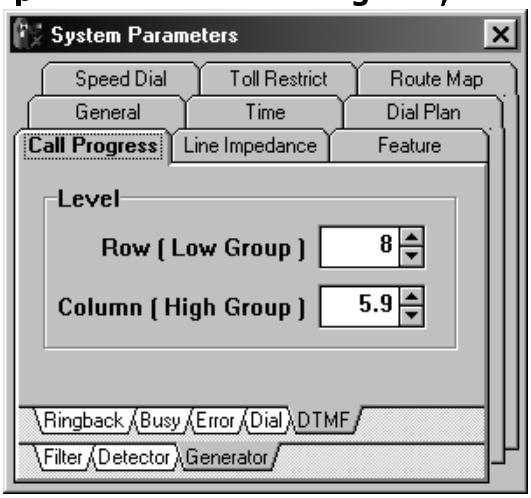


Table 6 - 34 System parameters: Call Progress, Generator, DTMF

DTMF Level		
Amplitude levels for the low and high frequencies associated with the DTMF digits 0-9, *, and #.	0-70 dB	Row (Low Group) 8 * Column (High Group) 5.9 *

* = default settings

6.21.21.1 DTMF digits and their respective frequencies

The following table illustrates the frequency associated with each DTMF digit. The column/high contains 4 frequencies (1209, 1336, 1477 and 1633). The row/ low contains 4 frequencies (697, 770, 852, 941).

Table 6 - 35

	1209 Hz	1336 Hz	1477Hz	1633 Hz
697 Hz	1	2	3	A
770 Hz	4	5	6	B
852 Hz	7	8	9	C
941 Hz	*	0	#	D

6.21.22 System parameters: Line Impedance

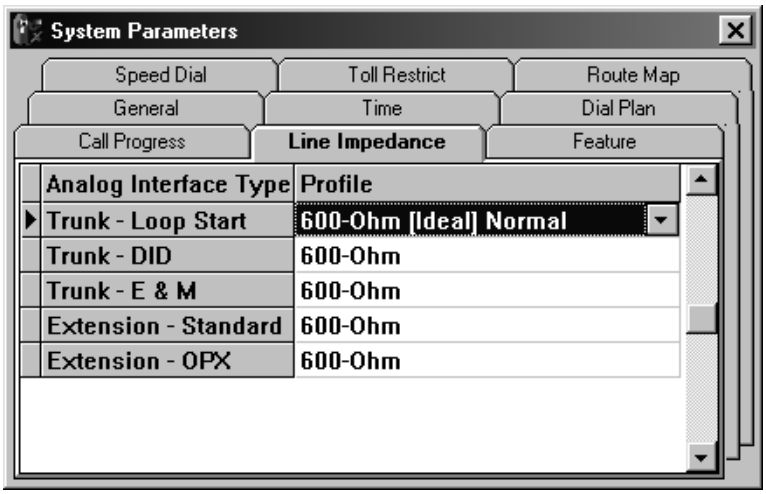


Table 6 - 36 System parameters: Line Impedance

Analog Interface Type	
Type of Analog Interface (e.g.,Loop Start).	
Profile	
Impedance profile for the analog interface. ***Note***: Line impedance profiles can only be configured through software configuration. Should the line impedance profile provided not meet the needs at a particular installation, e-mail a request for additional profiles at bbstechs@bbstelecom.com or contact a support representative.	600-Ohm *

* = default settings

6.21.23 System parameters: Feature, Assignment

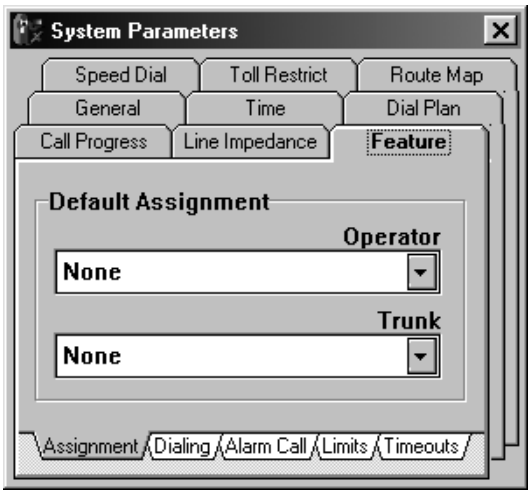


Table 6 - 37 System parameters: Feature, Assignment

Default Assignment-Operator	
Connect to for system-dialed calls placed to the operator or calls made by unassigned extensions. ***Note***: System-dialed calls are calls made by the Integrated Voice Processor as in the case where a caller presses **0** to reach the system operator while in a user's voice mailbox.	None * User ID User Group ID AutoAttendant ID Voice ID
Default Assignment-Trunk	
Trunk or Trunk Group utilized for system-dialed calls or unassigned extension calls.	None * Trunk ID Trunk Group ID

* = default settings

6.21.24 System parameters: Feature, Dialing

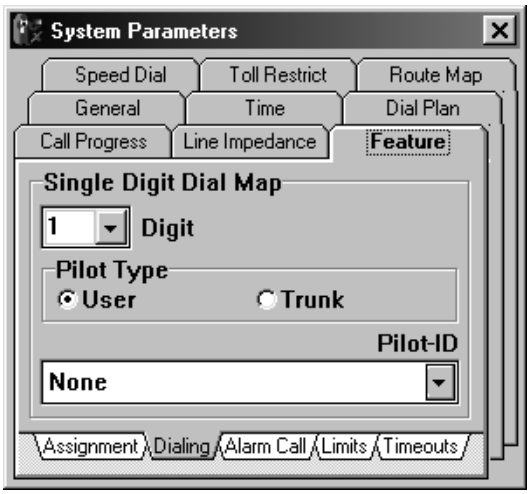


Table 6 - 38 System parameters: Feature, Dialing

Single Digit Dial Map: Digit	
Single digit access for users, user groups, trunks, trunk groups, auto attendant IDs, or voice IDs. E.g., If **3** is mapped to User 102, pressing the single digit **3** would connect to User 102.	0-9, #,*
Single Digit Dial Map: Pilot Type	
The type of entity to which the digit is mapped.	User /Trunk
Pilot-ID	
The ID to which the digit is mapped. ***Note:*** Plexus systems process a single digit at a time. If **3** is mapped to a user, pressing it will immediately connect you to the user. Therefore, it is not possible to connect to a user group or other entity with an ID of 300. Review the defined flexible numbering plan and access digits before mapping single digits.	None * User ID User Group ID Trunk ID Trunk Group ID AutoAttendant ID Voice ID

* = default settings

6.21.25 System parameters: Feature, Alarm Call

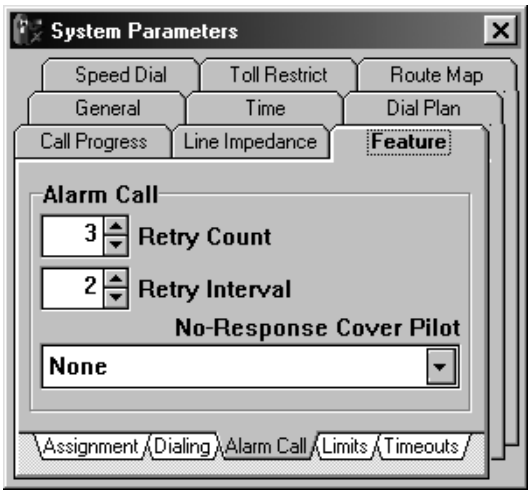


Table 6 - 39 System parameters: Feature, Alarm Call

Alarm Call: Retry Count		
The number of attempts the system will make to reach a user for either an appointment reminder or an alarm clock call.	0-255	3 *
Alarm Call: Retry Interval		
The period of time between retries.	0-255 minutes	2 minutes *
Alarm Call: No-Response Cover Pilot		
Connect to when no response is received for an alarm clock call. ***Note:*** The No-Response Cover Pilot does not apply to appointment reminders. Refer to chapter 18, "Hotel / Motel Package".	None * User ID User Group ID AutoAttendant ID Voice ID	

* = default settings

6.21.26 System parameters: Feature, Limits

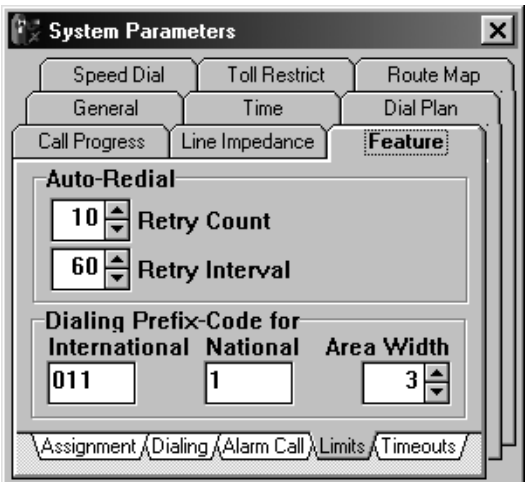


Table 6 - 40 System parameters: Feature, Limits

Auto Redial: Retry Count		
The number of attempts the system will make in completing an automatic redial.	0-255	10 Attempts
Auto-Redial: Retry Interval		
The period of time between retries.	0-255	60 seconds *
Dialing Prefix-Code: International		
The prefix the system will dial for an international call when a user utilizes Caller ID Redial (see “Caller ID Redial” - section 8.57).	numeric, 0-4 digits	011*
Dialing Prefix-Code: National		
The prefix the system will dial for a national call when a user utilizes Caller ID Redial (see “Caller ID Redial” - section 8.57).	numeric, 0-4 digits	1*
Dialing Prefix-Code: Area Width		
The number of digits in the system’s local area code (see “Caller ID Redial” - section 8.57).	0-10	3*

* = default settings

6.21.27 System parameters: Feature, Timeouts

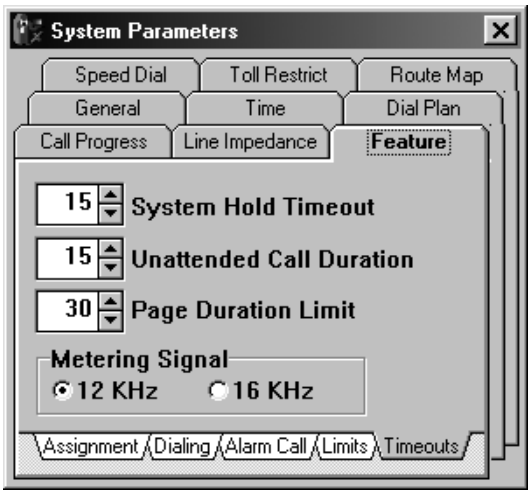


Table 6 - 41 System parameters: Feature, Timeouts

System Hold TImeout		
The period of time that a call will remain on System Hold before recalling the responsible user.	0-255 minutes	15 minutes*
Unattended Call-Duration		
The period of time that an unattended trunk call can go unsupervised before the call is automatically terminated. ***Note***: Unattended calls include unsupervised conferences, externally forwarded or diverted calls, and doorphone calls to an external number.	0-255 minutes	15 minutes*
Page Duration Limit		
The maximum duration of a page. The page is automatically terminated upon exceeding the duration limit.	0-255 seconds	30 seconds*
Metering Signal		
Frequency of the central office metering signal. ***Note***: Metering is available only on trunk cards with metering capabilities and in areas with metering service available from local phone service provider.	12KHz* or 16KHz	

* = default settings

6.22 Configure users

6.22.1 Configure Users

1. Ensure that all users have been created.
2. Click on the **User** tab to display the Users and User Group dialog window.
3. Double-click on each User in the User window.
4. Make appropriate entries and selections on each of the tabs.

Each user has a default system profile that can be modified to limit or expand the user's class of service.

Tip
To alleviate the repetitive tasks associated with configuring multiple users, right-click on a setting or field and select **Replicate ALL** or **Replicate Selected** [Replicate Selected is only available if users have been selected]. The Replicate options automatically assign a setting to all or selected users.

6.22.2 User: General

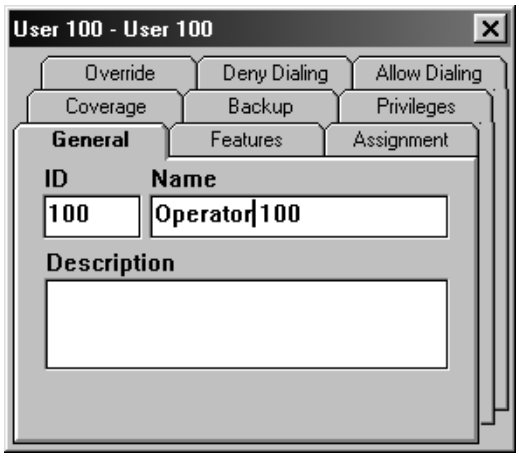


Table 6 - 42 User: General

ID	
Logical identifier for the user. ***Note:*** The ID is automatically assigned based on the User ID Sequence entered when the first user was created ***Note:*** To assign the user ID to a port, see “Extension port: General” - section 6.18.2.	Numeric Up to 4 digits
Name	
Informal identifier for the user. E.g., Heather.	Alphanumeric Up to 20 characters
Description	
Informal description or notes about the user. E.g., Receptionist	Alphanumeric

An extension port assigned to a user serves as the user's home station. It is not necessary to assign a user to a port. Users with no port assignment are referred to as *transient* users. Additional User IDs may also be given to a user to serve as a user alias for purposes of receiving different types of calls at a single station.

6.22.3 User: Features, Settings

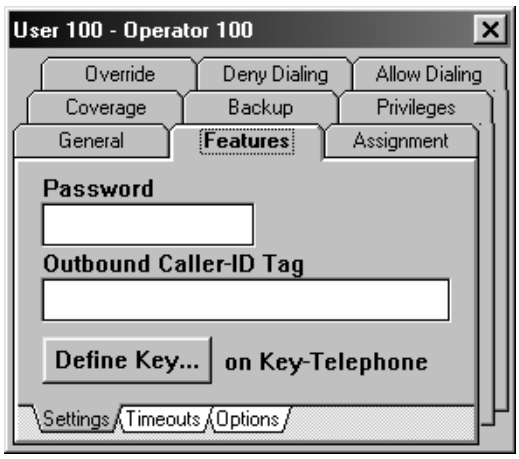


Table 6 - 43 User: Features, Settings

Password	
Optional log-in password. See "Password Security" - section 8.107.	Numeric, Up to 12 digits
Define Key… on Key-Telephone	
Press the **Define Key** button to define the programmable keys on the user's Plexus Key Telephone (if available).	

6.22.3.1 User: Features, Settings, Define Key

Unlike user privileges, programmable keys are assigned to a key telephone and not the user. The definitions are actually stored in memory that resides on the key telephone. A user's log-in privileges always override the physical key telephone's dialing or feature programming.

Notes
Only Plexus Key Telephones with firmware v1.01 or greater may be programmed through Plexus Administrator. The firmware version appears when the telephone is initializing.

To initialize a key telephone, dial **F99**. If the screen displays S/W 1_00 it cannot be programmed using Plexus Administrator, but must be programmed at the key telephone.

To define the programmable keys on a Plexus Key Telephone, proceed as follows:

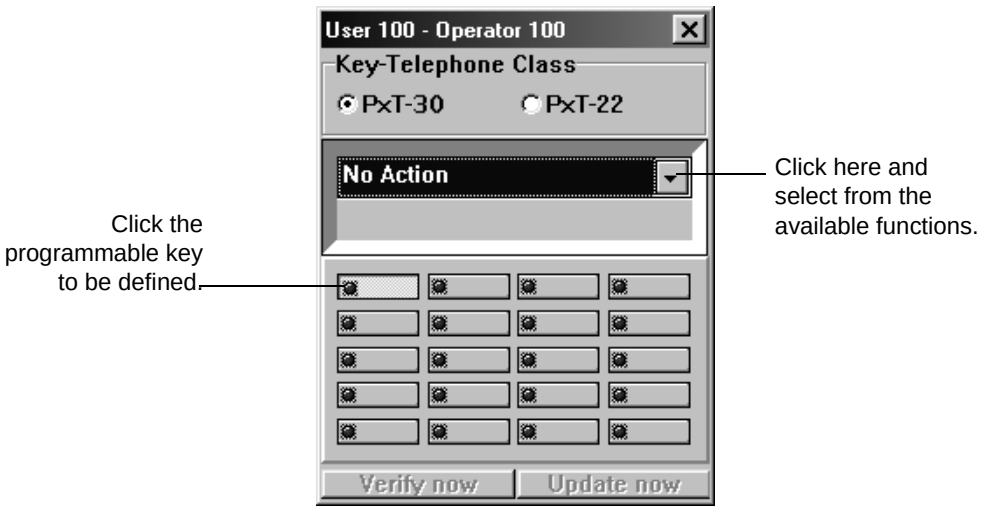
1 Click on the **User** tab.

2 Click on the **Feature** tab.

3 Click on the **Define Key** button.

Templates will appear for the various models of Plexus Key Telephones.

4 Click on the tab that corresponds to the available model. The following dialog box will appear:



5 Click on the programmable key to be defined.

The selected key will become highlighted.

6 Click on the options arrow, and select from the listed functions, as seen in the table below.

7 Programmable key definitions can be replicated to other key telephones on the system. To replicate a key definition to all other key phones on the system, press the options arrow, right click the mouse, and then choose **Replicate All**.

8 To replicate selected users: use the **CTRL** key to select users from the list of users. Then, press the options arrow and choose **Replicate Selected**.

Note
To see the programming already on a user's Key Telephone, Press **Verify now** after linking with the system. This feature works with version 2.0 Key Telephones and above only.

After programming the keys, you will need to send the parameters to the key telephones. See "Updating key telephones" - section 6.22.3.3.

6.22.3.2 Available functions for the Plexus Key Telephone

Table 6 - 44 User: Features, Settings

No Action*
Instructs the system to not overwrite key telephone programming during an update.
Undefined
Not programmed.
Map/Monitor Users
Monitors the status of a system user. E.g., When the monitored user goes off hook, the mapped key will illuminate.
Map/Monitor Trunks
Monitors the status of a trunk. E.g., When the monitored trunk is in use, the mapped key will illuminate.
Function
Activates a function [See "Feature Reference" on page 1 for a full description of the function codes]. Call Record is function 1, and Live Call Screening is function 2.
Feature
Activates a feature [See "Overview of Features" on page 1 for features codes].
Speed-Dial
Dials a frequently used telephone number. Commas provide a 2 second pause.
Default
Several buttons on version 2.0 Key Telephones have additional features that can be programmed. See User's Guide - Programming Keyphone for options.

* = default settings

6.22.3.3 Updating key telephones

To update the key telephone version 1.01, proceed as follows:

1. After linking with the system, click on the **Update** menu at the top of the Administrator program.
2. Click on **Key-Phone**. This will update all version 1.01 and above Key Telephones physically attached to the system.

To update the key telephone version 2.0 and above, proceed as follows:

1. After linking with the system, click on the **Update now** button on the screen. This will only update the phone at that user's station.

or

1. After linking with the system, click on the **Update** menu at the top of the Administrator program.
2. Click on **Key-Phone**. This will update all version 1.01 and above Key Telephones physically attached to the system.

6.22.4 User: Features, Timeouts

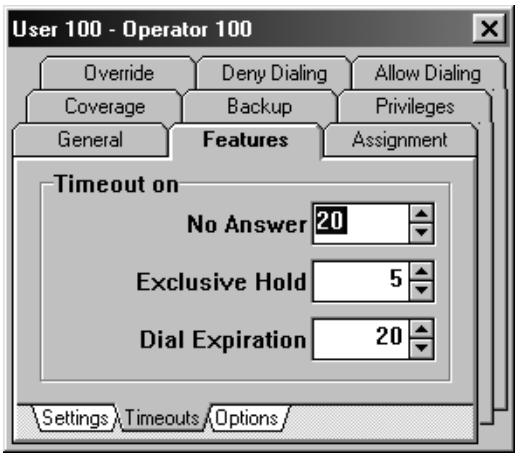


Table 6 - 45

Timeout on No Answer		
The period of time before an unanswered call routes to the user's backup or coverage. ***Note:*** See"Call Backup - User / User Group" - section 8.27 and "Call Coverage" - section 8.32.	0-255 seconds	20 seconds*
Timeout on Exclusive Hold		
The period of time that a call will remain on Exclusive Hold before recalling the responsible user. ***Note:*** See "Call Hold: Exclusive" - section 8.37 for further recall interactions.	0-255 minutes	5 minutes*
Timeout on Dial Expiration		
The amount of time that an off-hook station receives internal dial tone before receiving error tone.	0-255 seconds	20 seconds*

* = default settings

6.22.5 User: Features, Options

Table 6 - 46 User: Features, Options

Auto Log-In	
Enables the user to be automatically logged-in each time the system boots.	Enable*/Disable
Auto Cover on Internal Calls	
Enables an automated version of the call cover feature for the user. ***Note***: A user with Auto Cover enabled will automatically route to the applicable coverage entity when calling an unavailable user.	Enable/Disable*
Validate Trunk Access during Day	
The Plexus system will only allow the user to place an external call after the user verifies the user's trunk access privileges while the system is in the day mode. See "Authorize User" - section 8.12.	Enable/Disable*
Validate Trunk Access at Night	
The Plexus system will only allow the user to place an external call after the user verifies the user's trunk access privileges while the system is in the night mode. See "Authorize User" - section 8.12.	Enable/Disable*
Forced Account Code	

Table 6 - 46 User: Features, Options (continued)

The Plexus system will only allow the user to place an external call after the user enters an account code for the call. See “Account Codes: Forced” - section 8.4.	Enable/Disable*
Least Cost Routing Enabled	
The Plexus system will route external calls based on information entered in the Least Cost Routing table. See chapter 14, "Least Cost Routing.	Enable/Disable*
Block on Call Termination	
Prevents an internal dial tone from being sent to a phone after disconnecting from a phone call. See “Block On Call Termination” - section 8.19.	Enable/Disable*

* = default settings

6.22.6 User: Assignment, Group-Access

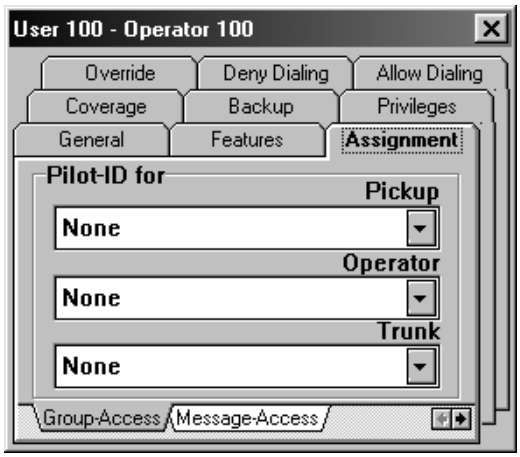


Table 6 - 47 User: Assignment, Group-Access

Pilot-ID for Pickup	
Pickup Group or **Pickup User** that the user picks up when using **F07** or **F08.** See “Call Pickup: Intra-Group” - section 8.44.	None * User ID User Group ID AutoAttendant ID Voice ID
Pilot-ID for Operator	
Connect to for calls to the operator. E.g., when the user presses **0**.	None * User ID User Group ID AutoAttendant ID Voice ID
Pilot-ID for Trunk	
Trunk or trunk group utilized by the user for external outbound calls. This parameter also determines which trunks will be used for automated off-site dialing (e.g., External Forward and Remote Call Notification) for the user.	None * Trunk ID Trunk Group ID

* = default settings

6.22.7 User: Assignment, Message-Access

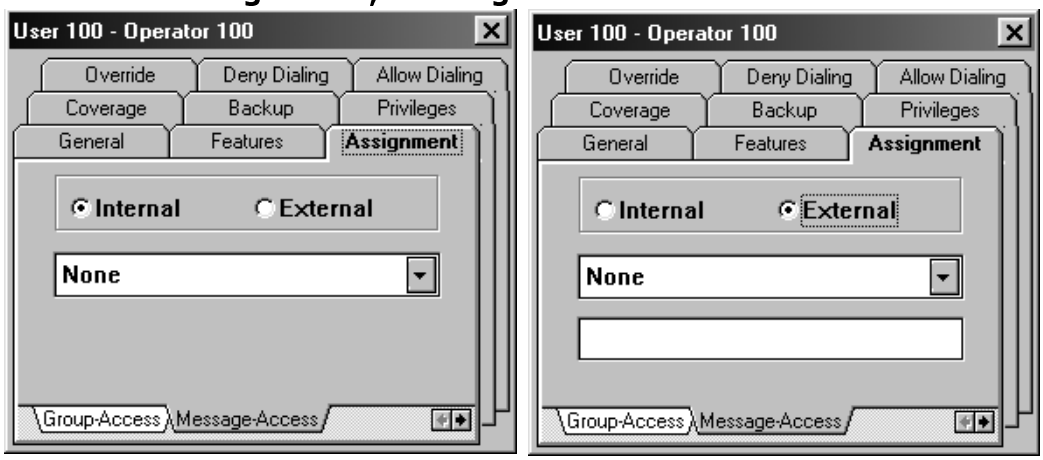


Table 6 - 48 User: Assignment, Message-Access

Message-Access: Internal	
System entity that the user accesses when checking messages. ***Note:*** On systems with an Integrated Voice Processor, this is set to a Voice ID. In the absence of an Integrated Voice Processor, this can be set for a receptionist, operator, or front desk where messages are taken manually.	None* User ID User Group ID Voice ID
Message-Access: External	
Trunk / Trunk Group that the user accesses when checking messages. ***Note:*** On systems with an Integrated Voice Processor, this is set to an Internal Voice ID. In the absence of an Integrated Voice Processor, this can be set for an answering service, etc.	None* Trunk ID Trunk Group ID
Message-Access: External - Dial Number	
Externally dialed telephone number that the system dials for the user to check messages.	Numeric, 0-9 up to 20 digits

* = default settings

6.22.8 User: Assignment, Off-Hook Preference, User

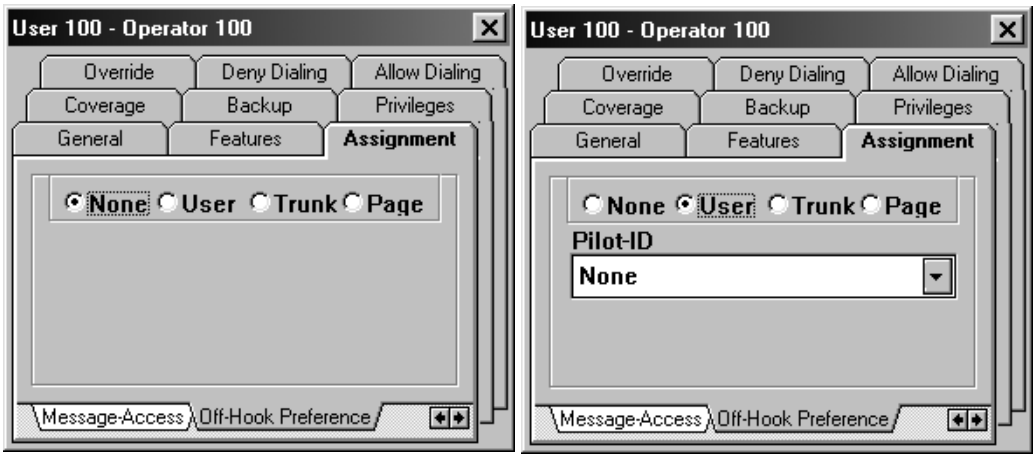


Table 6 - 49 User: Assignment, Off-Hook Preference, User

Pilot-ID	
Internal **connect to** initiated by going off-hook. E.g., A hotel lobby phone which automatically calls the hotel switch operator when someone picks up the handset.	None * User ID User Group ID AutoAttendant ID Voice ID

* = default settings

6.22.9 User: Assignment, Off-Hook Preference, Trunk

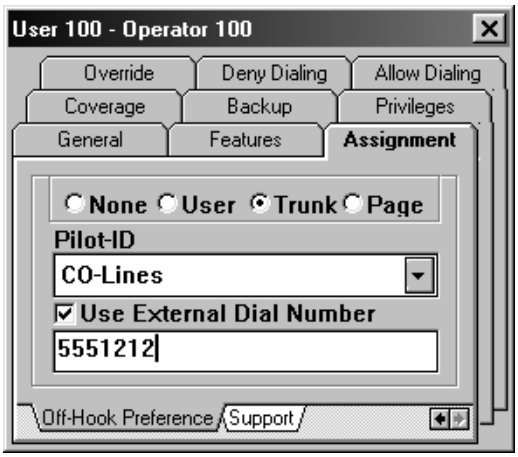


Table 6 - 50 User: Assignment, Off-Hook Preference, Trunk

Pilot-ID	
Trunk or trunk group the user is automatically connected to by going off-hook. e.g., An elevator phone which automatically accesses an outside line when someone picks up the handset, so that emergency numbers can be dialed.	None * Trunk ID Trunk Group ID
Use External Dial Number	
Telephone number associated with the trunk off-hook preference ***Note:*** When enabled, the telephone number is automatically dialed after the user is connected to the trunk or trunk group designated in the trunk off-hook preference Pilot-ID above. E.g., An elevator phone could be set to dial 911 when someone lifts the handset.	Enable/Disable + Phone #

* = default settings

6.22.10 User: Assignment, Off-Hook Preference, Page

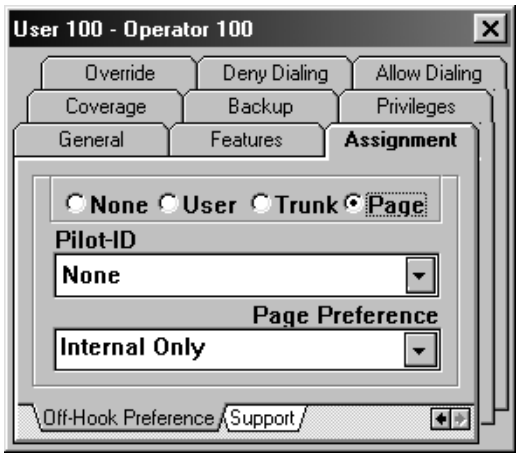


Table 6 - 51 User: Assignment, Off-Hook Preference, Page

Pilot-ID	
Paging service **connect to** initiated by going off-hook. e.g., going off-hook performs an internal page to the selected user group ***Note:*** The Pilot-ID applies only to internal pages. To page all users, do not select a Pilot-ID and choose internal for the page preference parameter below. ***Note:*** Page refers to the **Page** features.	None * User Group ID
Page Preference	
The type of paging service to be utilized.	Internal * External Answer

* = default settings

6.22.11 User: Assignment, Support

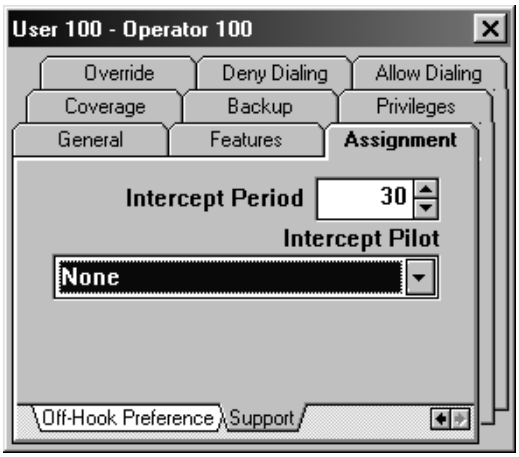
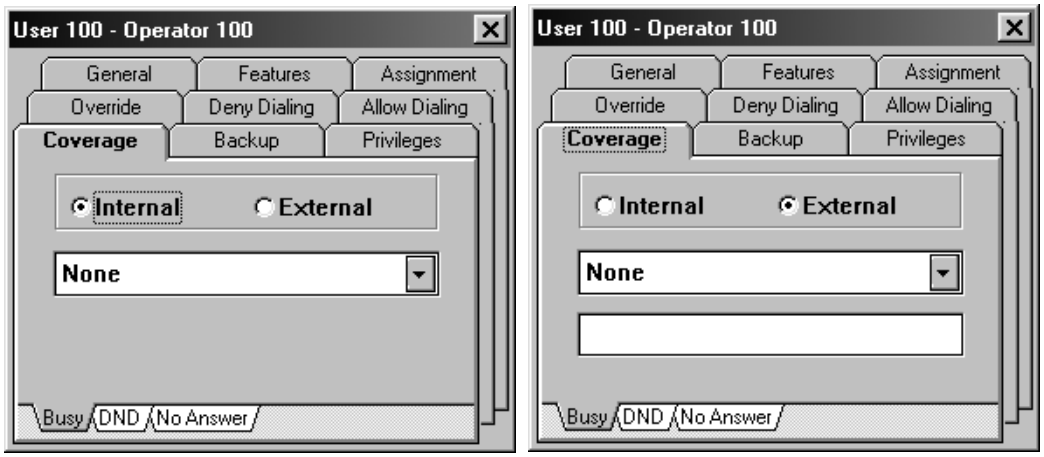


Table 6 - 52 User: Assignment, Support

Intercept Period		
The period of time between connections to the intercept pilot while a caller is waiting in a user's exclusive hold queue. ***Note:*** See "Intercept Service" - section 8.84.	0-255 seconds	30 seconds*
Intercept Pilot		
Intercept **connect to** for calls in queue for the user.	None * User ID User Group ID Auto Attendant ID Voice ID	

*= default settings

Note
Coverage entities (i.e., Pilot-IDs) should be chosen carefully as calls routed to an unavailable entity are dropped.



Coverage Pilot IDs will only be used if the user's Backup Pilot IDs are unavailable or not defined.

Note
In Coverage, if a Voice ID is set for the Pilot ID, the call will route to the user's voice mailbox. For the call to route to the Auto Attendant, the user's Backup must be set to the Voice ID.

Table 6 - 53 User: Coverage, Busy

Pilot-ID on Busy: Internal	
Where the call will be transferred to when the user is busy.	None * User ID User Group ID AutoAttendant ID Voice ID (see above note)
Pilot-ID on Busy: External	
Trunk or trunk group the call will be transferred to when the user is busy. An external number must be defined or the call will be terminated. (This is used to route the calls to an external entity; e.g., answering service.)	None * Trunk ID Trunk Group ID

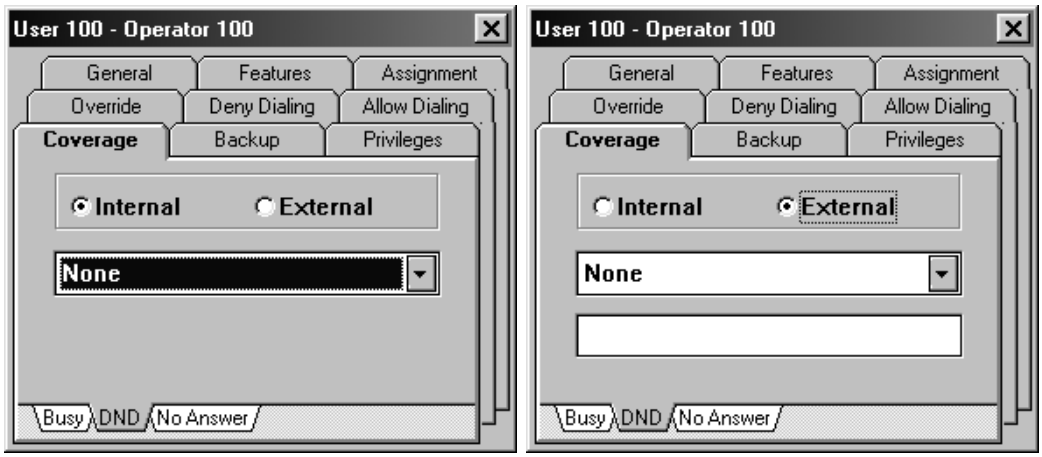
7DEOH 8VHU &RYHUDJH %XV\

External Dial Number	
Externally dialed number for External Coverage.	none*, 0-9

* = default settings

8VHU & RYHUDJH '1'

Note
Coverage entities (i.e., Pilot-IDs) should be chosen carefully as calls routed to an unavailable entity are dropped.



Coverage Pilot IDs will only be used if the user's Backup Pilot IDs are unavailable or not defined.

Note
In Coverage, if a Voice ID is set for the Pilot ID, the call will route to the user's voice mailbox. For the call to route to the Auto Attendant, the user's Backup must be set to the Voice ID.

7DEOH 8VHU &RYHUDJH '1'

Pilot-ID on Busy: Internal	
Where the call will be transferred to when the user is in DND.	None * User ID User Group ID AutoAttendant ID Voice ID (see above note)
Pilot-ID on Busy: External	
Trunk or trunk group the call will be transferred to when the user is in DND. An external number must be defined or the call will be terminated. (This is used to route the calls to an external entity; e.g., answering service.)	None * Trunk ID Trunk Group ID

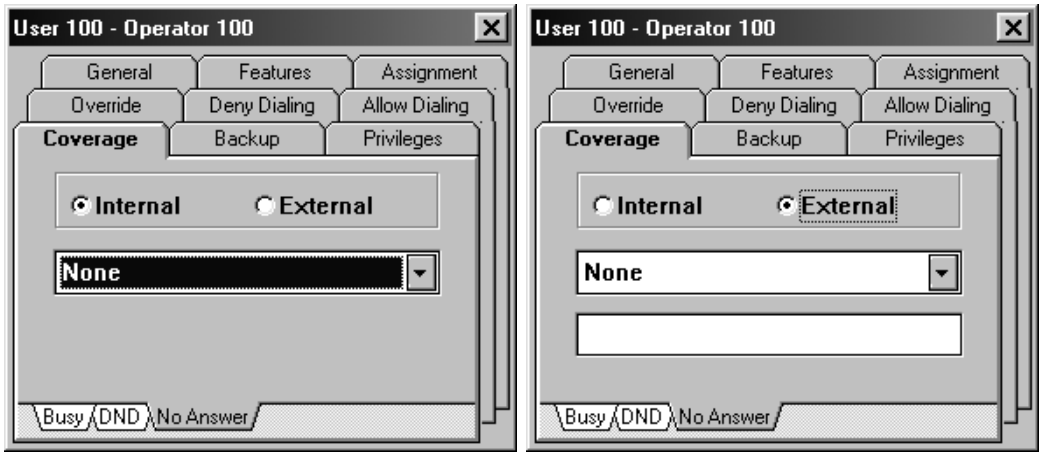
7DEOH 8VHU &RYHUDJH '1'

External Dial Number	
Externally dialed number for External Coverage.	none*, 0-9

* = default settings

8VHU & RYHUDJH 1R $QVZHU

Note
Coverage entities (i.e., Pilot-IDs) should be chosen carefully as calls routed to an unavailable entity are dropped.



Coverage Pilot IDs will only be used if the user's Backup Pilot IDs are unavailable or not defined.

Note
In Coverage, if a Voice ID is set for the Pilot ID, the call will route to the user's voice mailbox. For the call to route to the Auto Attendant, the user's Backup must be set to the Voice ID.

7DEOH 8VHU &RYHUDJH 1R $QVZHU

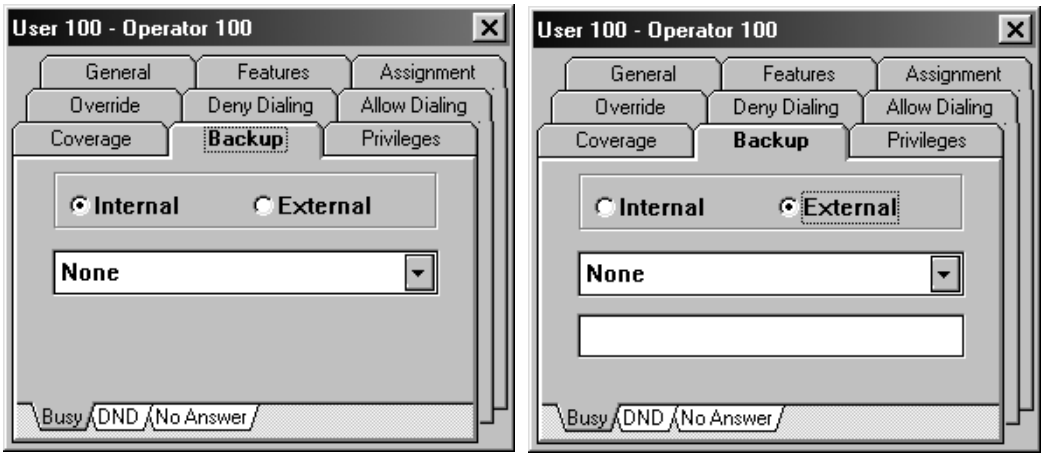
Pilot-ID on Busy: Internal	
Where the call will be transferred to when the user does not answer.	None * User ID User Group ID AutoAttendant ID Voice ID (see above note)
Pilot-ID on Busy: External	
Trunk or trunk group the call will be transferred to when there is no answer. An external number must be defined or the call will be terminated. (This is used to route the calls to an external entity; e.g., answering service.)	None * Trunk ID Trunk Group ID

7DEOH 8VHU &RYHUDJH 1R $QVZHU

External Dial Number	
Externally dialed number for External Coverage.	none*, 0-9

* = default settings

8VHU %DFNXS %XV\



If a Backup entity is unavailable (Busy, DND, No Answer) or is not defined, the call will follow the user's coverage.

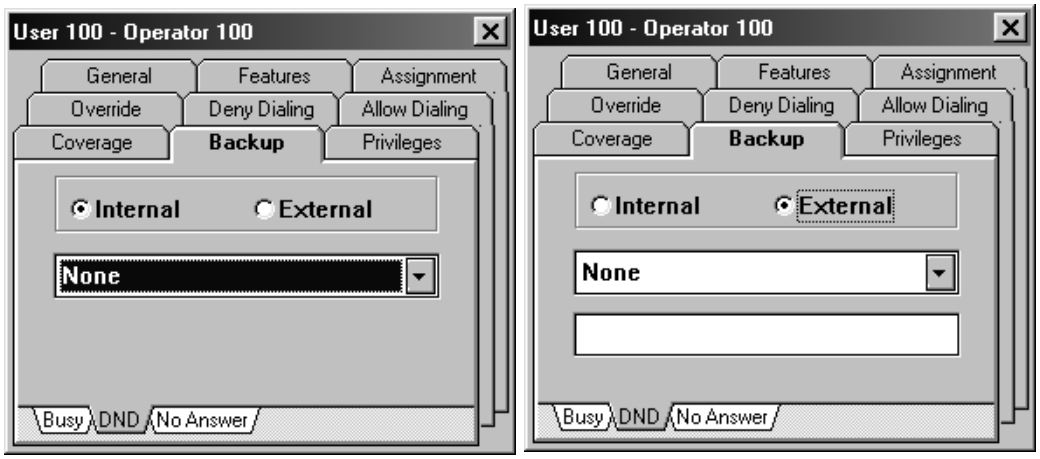
Note
In Backup, if a Voice ID is set for the Pilot ID, the call will route to the Auto Attendant. For the call to route to the user's voice mailbox, the user's Coverage must be set to the Voice ID.

7DEOH 8VHU %DFNXS %XV\

Pilot-ID on Busy: Internal	
Where the call will be transferred to when the user is busy.	None * User ID User Group ID AutoAttendant ID Voice ID (see above note)
Pilot-ID on Busy: External	
Trunk or trunk group the call will be transferred to when the user is busy. (This is used to route the calls to an external entity; e.g., answering service.)	None * Trunk ID Trunk Group ID
External Dial Number	
Externally dialed number for External Backup.	none*, 0-9

* = default settings

8VHU %DFNXS '1'



If a Backup entity is unavailable (Busy, DND, No Answer) or is not defined, the call will follow the user's coverage.

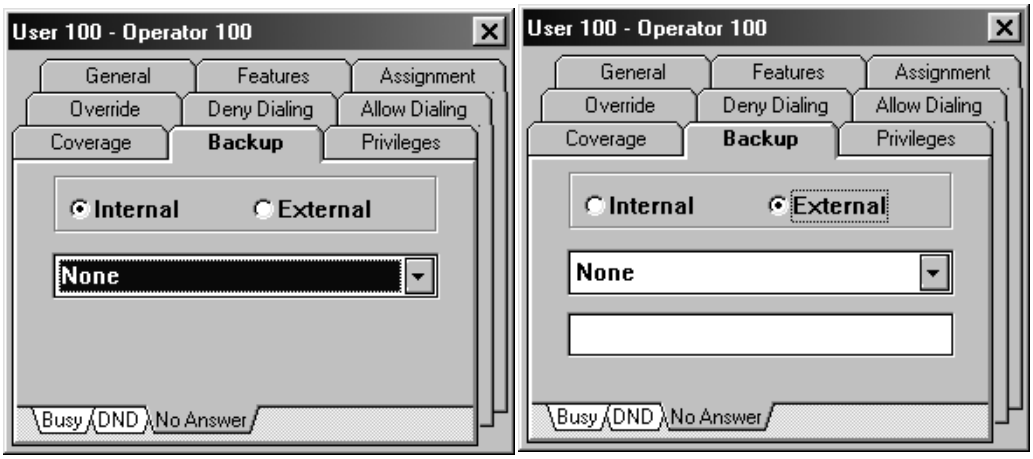
Note
In Backup, if a Voice ID is set for the Pilot ID, the call will route to the Auto Attendant. For the call to route to the user's voice mailbox, the user's Coverage must be set to the Voice ID.

7DEOH

Pilot-ID on Busy: Internal	
Where the call will be transferred to when the user is in DND.	None * User ID User Group ID AutoAttendant ID Voice ID (see above note)
Pilot-ID on Busy: External	
Trunk or trunk group the call will be transferred to when the user is in DND. (This is used to route the calls to an external entity; e.g., answering service.)	None * Trunk ID Trunk Group ID
External Dial Number	
Externally dialed number for External Backup.	none*, 0-9

* = default settings

8VHU %DFNXS 1R $QVZHU



If a Backup entity is unavailable (Busy, DND, No Answer) or is not defined, the call will follow the user's coverage.

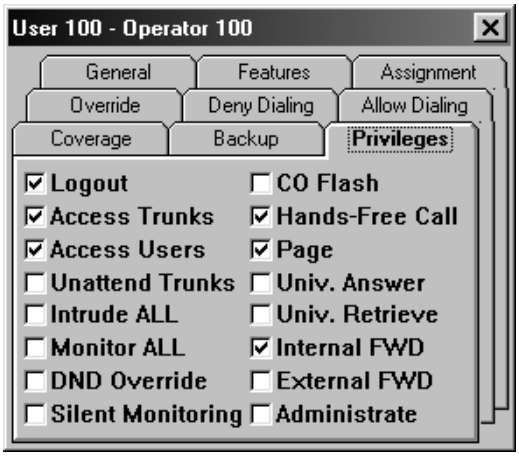
Note
In Backup, if a Voice ID is set for the Pilot ID, the call will route to the Auto Attendant. For the call to route to the user's voice mailbox, the user's Coverage must be set to the Voice ID.

7DEOH 8VHU %DFNXS 1R $QVZHU

Pilot-ID on Busy: Internal	
Where the call will be transferred to when the user does not answer.	None * User ID User Group ID AutoAttendant ID Voice ID (see above note)
Pilot-ID on Busy: External	
Trunk or trunk group the call will be transferred to when the user does not answer. (This is used to route the calls to an external entity; e.g., answering service.)	None * Trunk ID Trunk Group ID
External Dial Number	
Externally dialed number for External Backup.	none*, 0-9

* = default settings

8VHU 3ULYLOHJHV



7DEOH 8VHU 3ULYLOHJHV

Logout	
Permits the user to log out. ***Note:*** Without the logout privilege, the user is tied to a particular station as on a conventional telephone system.	Enable */ Disable
Access Trunks	
Permits the user to access outside lines.	Enable */ Disable
Access Users	
Permits the user to access other users (i.e., make internal calls).	Enable */ Disable
Unattend Trunks	
Permits the user to transfer one external call to another external call (“trunk-to-trunk transfer”) and create an unsupervised conference. ***Note***: All unattended calls are subject to the Unattended Call-Duration timeout. See “System parameters: Feature, Timeouts” - section 6.21.27.	Enable / Disable *

7DEOH 8VHU 3ULYLOHJHV FRQWLQXHG

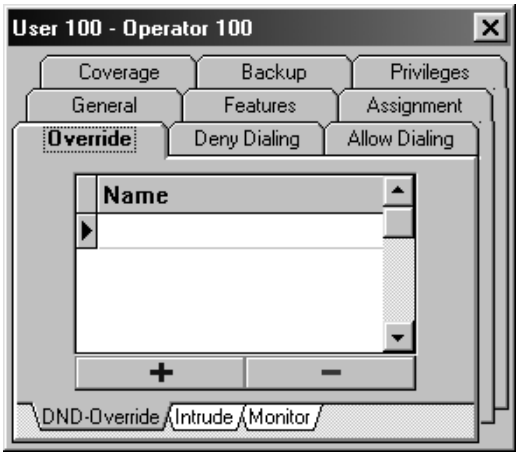
Intrude ALL	
Permits the user to intrude on ALL busy users and trunks. All parties on the call will hear a stutter tone to indicate that someone has intruded on the call. Plexus Key Telephone users (display model) will see “conferenced” on their LCD. ***Note:*** Intrude privileges may be limited by listing specific users [See “User: Override, Intrude” - section 6.22.20] rather than granting the privilege universally.	Enable / Disable *
Monitor ALL	
Permits the user to monitor ALL busy users and trunks. All parties on the call will hear a stutter tone to indicate that someone has intruded on the call. Plexus Key Telephone users (display model) will see “conferenced” on their LCD. ***Note:*** Monitor privileges may be limited by listing specific users [See “User: Override, Monitor” - section 6.22.21] rather than granting the privilege universally.	Enable / Disable *
DND Override	
Permits the user to override the Do Not Disturb (DND) mode of ALL users. ***Note:*** DND override privileges may be limited by listing specific users [See “User: Override, DND Override” on page 95] rather than granting the privilege universally.	Enable / Disable *
Silent Monitoring	
Permits a user to silently monitor ALL busy users and trunks or selected users and trunks as defined by the User: Override, Monitor list. ***Note***: Silent Monitoring provides no indication to the busy user or the other party that the call is being monitored.	Enable / Disable *
CO Flash	
Permits the user to send a ‘Flash’ to the CO to utilize Centrex features.	Enable / Disable *
Hands-Free Call	
Permits the user to place hands-free calls to other users.	Enable * / Disable

7DEOH 8VHU 3ULYLOHJHV FRQWLQXHG

Page	
Permits the user to issue internal and external pages. ***Note:*** Page refers to the Paging Service feature.	Enable */ Disable
Univ Answer	
Permits the user to answer any ringing user, user group, or trunk on the system [See “Universal Answer” - section 8.134].	Enable/ Disable *
Univ Retrieve	
Enables a user to retrieve an internal or external call from another user’s exclusive hold queue or from a user group’s busy hold queue. ***Note:*** Universal Retrieve is not available with Key Telephone firmware version 1.01 or previous versions.	Enable / Disable *
Internal FWD	
Permits the user to forward his/her calls to another user on the system. [See “Call Follow-Me” - section 8.34 and “Call Forwarding” - section 8.35]	Enable */ Disable
External FWD	
Permits the user to forward his/her calls to an off-site telephone [See “Call Forwarding” - section 8.35].	Enable/Disable *
Administrate	
Permits the user to activate System Administrator privileges [See “System Administrator Privileges (SAP)” - section 8.126].	Enable/Disable *

* = default settings

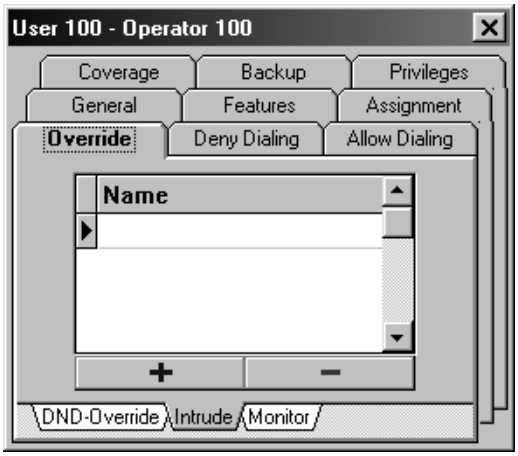
8VHU 2YHUULGH '1' 2YHUULGH



7DEOH 8VHU 2YHUULGH '1' 2YHUULGH

Name
Allows the user to override the Do Not Disturb (DND) mode of the listed users. User Name

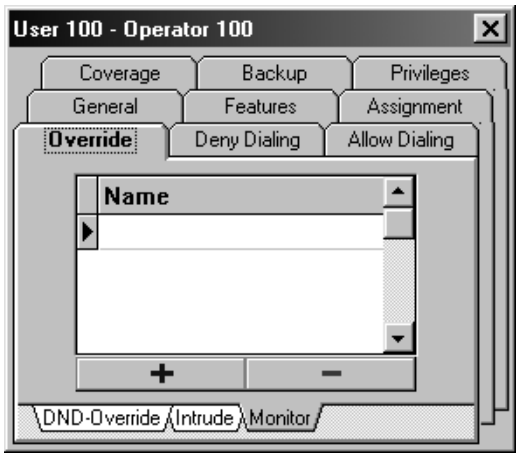
8VHU 2YHUULGH ,QWUXGH



7DEOH 8VHU 2YHUULGH ,QWUXGH

Name	
Allows the user to intrude on the listed users.	User Name

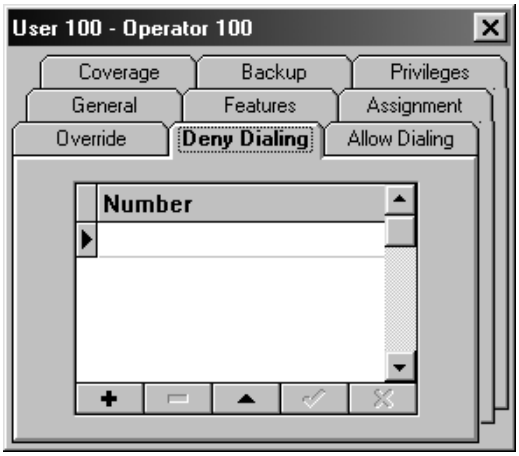
8VHU 2YHUULGH 0RQLWRU



7DEOH 8VHU 2YHUULGH 0RQLWRU

Name	
Allows the user to monitor the listed users.	User Name

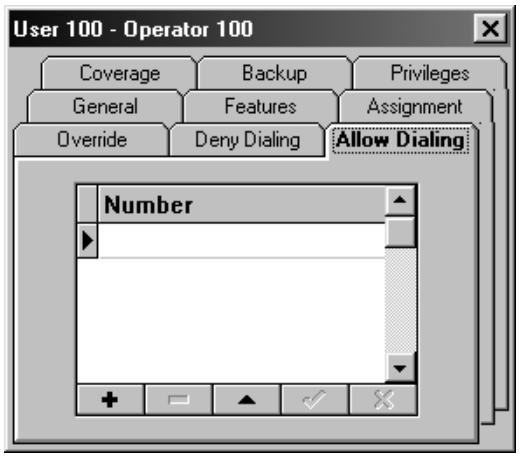
8VHU 'HQ\ 'LDOLQJ



7DEOH 8VHU 'HQ\ 'LDOLQJ

Number	
List of phone numbers and prefixes that the system will generally not allow the user to dial. Specific exceptions are listed in the user's allow dialing list. E.g., 1900 would terminate all calls, dialed by the user, with this prefix unless the specific telephone number is in the user's allow dialing list.	Phone # / Phone # prefix

8VHU $OORZ 'LDOLQJ



7DEOH 8VHU $OORZ 'LDOLQJ

Number	
List of phone numbers that the user is specifically allowed to dial. ***Note:*** This will override numbers entered in the Deny Dial table. E.g., '1' can be entered in the Deny Dial table to block all toll calling but '1800' can be entered in the Allow Dial table to allow toll free calling.	Phone # / Phone # prefix

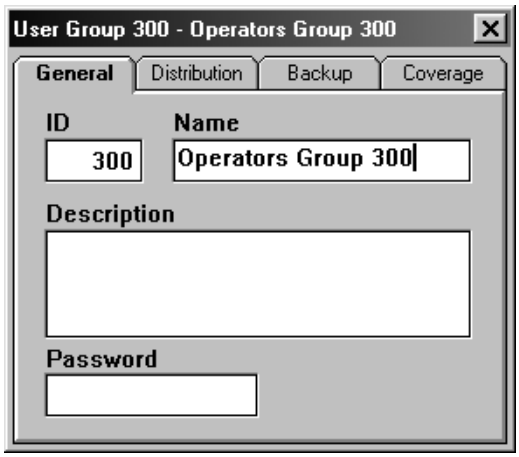
6.23 Configure user groups

1. Ensure that Users and User Groups have been created.
2. Click on the **User** tab to display the User and User Group dialog window.
3. Double-click on each user group in the User Group window.
4. Make appropriate entries and selections on each of the tabs.

Tip

To alleviate the repetitive tasks associated with configuring multiple user groups, right-click on a setting or field and select **Replicate**. Replicate automatically assigns a setting to all user groups.

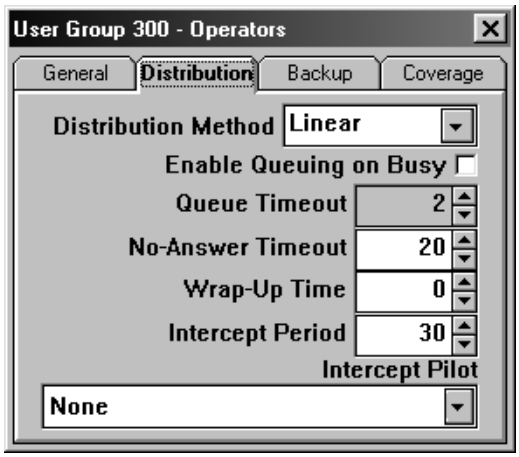
8VHU *URXS *HQHUDO



7DEOH 8VHU *URXS *HQHUDO

ID	
Logical identifier for the user group. ***Note:*** The ID is automatically assigned based on the User Group ID Sequence entered when the first user group was created.	Numeric Up to 4 digits
Name	
Informal identifier for the user group. E.g., Sales, Technical Support	Alphanumeric Up to 20 characters
Description	
Informal description or notes about the user group. E.g., The Sales group consists of all sales personnel.	Alphanumeric
Password	
Password that must be entered by users to Log On to a group [see “Agent Log Off / Agent Log On” - section 8.5.	Numeric

8VHU *URXS 'LVWULEXWLRQ



7DEOH 8VHU *URXS 'LVWULEXWLRQ

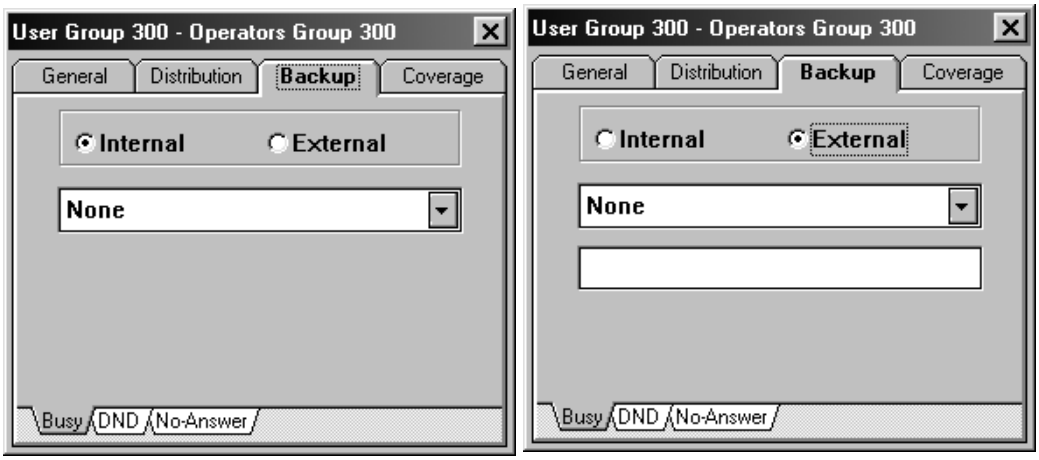
Distribution Method		
Method utilized in hunting for an available user in the user group. ***Note***: See “Call Distribution Management” - section 8.33 for detailed descriptions of distribution methods.	Linear* Circular Least Busy Multiple Manual Single Shot	
Enable Queuing on Busy		
Whether Busy Queuing is enabled for calls to this group. ***Note:*** See “Busy Queuing - User Group” - section 8.22.	Enable / disable*	
Queue Timeout		
The period of time that a call may remain in queue waiting for an available user before routing to backup and/or coverage. ***Note:*** See “Busy Queuing - User Group” - section 8.22.	0-255 minutes	2 minutes*
No Answer Timeout		
The period of time before an unanswered call rolls over to the next available user. If there is not a next available user, the call will route to the defined backup / coverage.	0-255 seconds	20 seconds*

7DEOH 8VHU *URXS 'LVWULEXWLRQ

Wrap Up Time		
The period of time allotted for a user to wrap-up after terminating a call. No additional calls will route to the user until the wrap-up time expires.	0-255 seconds	15 seconds*
Intercept Period		
Refers to the period of time between reassurance messages while a caller is waiting in the busy queue. ***Note:*** Enable/Disable *using check-box.	0-255 seconds	30 seconds*
Intercept Pilot ID		
System entity providing the audio prompts for call holding in the queue. ***Note:*** To provide a system default message or a prerecorded message, the IVP card must be properly set up. See “ Setting up Hold Queue Intercepts” in the IVP section.	None * User ID User Group ID AutoAttendant ID Voice ID	

* = default settings

8VHU *URXS %DFNXS %XV\



For a user group, busy is defined as when all users in the user group are busy or in DND with at least one agent in the busy state. If a Backup entity is unavailable (Busy, DND, No Answer) or is not defined, the call will follow the user group's coverage.

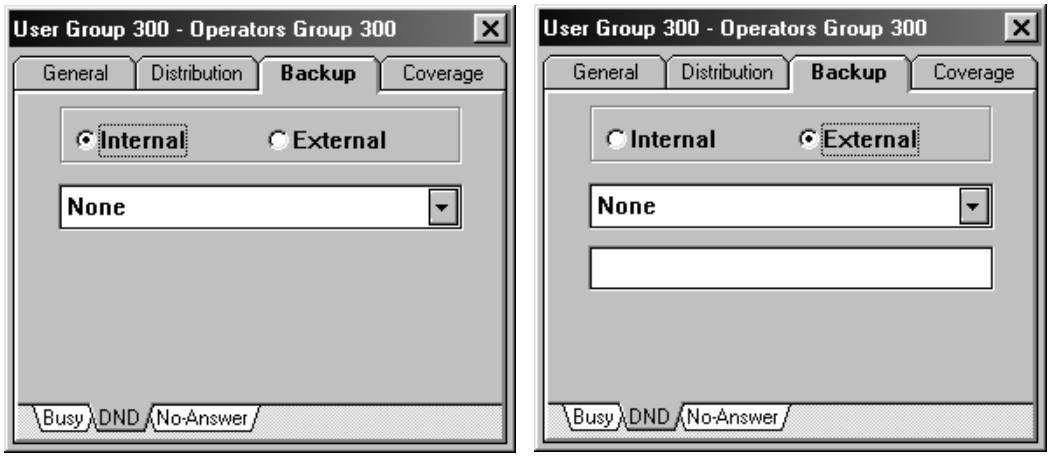
Note
In Backup, if a Voice ID is set for the Pilot ID, the call will route to the Auto Attendant. For the call to route to the user group's voice mailbox, the user group's Coverage must be set to the Voice ID.

7DEOH 8VHU *URXS %DFNXS %XV\

Pilot-ID on Busy: Internal	
Where the call will be transferred to when all users in the user group are busy or in DND (at least one agent must be in the busy state).	None * User ID User Group ID AutoAttendant ID Voice ID (see above note)
Pilot-ID on Busy: External	
Trunk or trunk group the call will be transferred to when the user group is busy. (This is used to route the calls to an external entity; e.g., answering service.)	None * Trunk ID Trunk Group ID
External Dial Number	
Externally dialed number for External Backup.	none*, 0-9

* = default settings

8VHU *URXS %DFNXS '1'



For a user group, DND is defined as when all users in the user group are in DND. If a Backup entity is unavailable (Busy, DND, No Answer) or is not defined, the call will follow the user group's coverage.

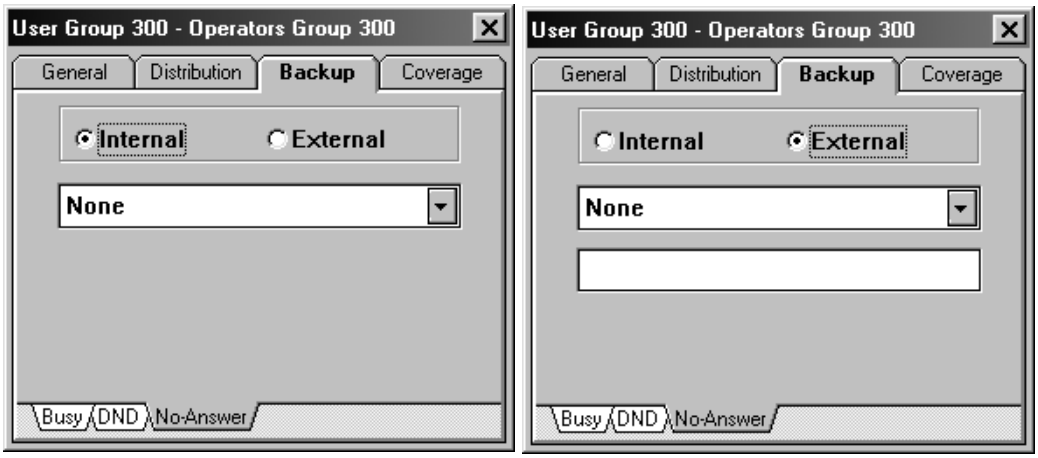
Note
In Backup, if a Voice ID is set for the Pilot ID, the call will route to the Auto Attendant. For the call to route to the user group's voice mailbox, the user group's Coverage must be set to the Voice ID.

7DEOH 8VHU *URXS %DFNXS '1'

Pilot-ID on Busy: Internal	
Where the call will be transferred to when all users in the user group are in DND.	None * User ID User Group ID AutoAttendant ID Voice ID (see above note)
Pilot-ID on Busy: External	
Trunk or trunk group the call will be transferred to when the user group is DND. (This is used to route the calls to an external entity; e.g., answering service.)	None * Trunk ID Trunk Group ID
External Dial Number	
Externally dialed number for External Backup.	none*, 0-9

* = default settings

8VHU *URXS %DFNXS 1R $QVZHU



For a user group, No Answer is defined as when at least one user is not busy or in DND and the call is not answered after ringing to the available user(s). If a Backup entity is unavailable (Busy, DND, No Answer) or is not defined, the call will follow the user group's coverage.

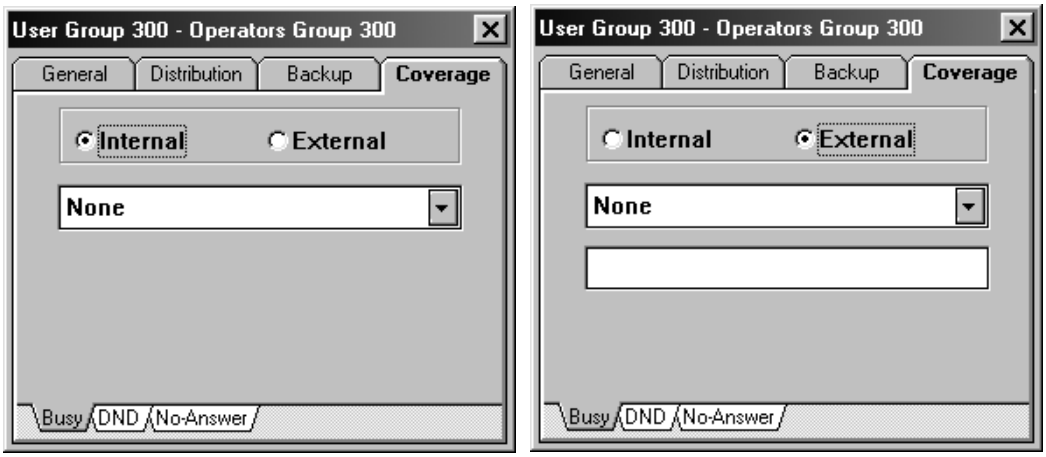
Note
In Backup, if a Voice ID is set for the Pilot ID, the call will route to the Auto Attendant. For the call to route to the user group's voice mailbox, the user group's Coverage must be set to the Voice ID.

7DEOH 8VHU *URXS %DFNXS 1R $QVZHU

Pilot-ID on Busy: Internal	
Where the call will be transferred to when there is no answer.	None * User ID User Group ID AutoAttendant ID Voice ID (see above note)
Pilot-ID on Busy: External	
Trunk or trunk group the call will be transferred to when there is no answer. (This is used to route the calls to an external entity; e.g., answering service.)	None * Trunk ID Trunk Group ID
External Dial Number	
Externally dialed number for External Backup.	none*, 0-9

* = default settings

8VHU *URXS & RYHUDJH %XV\



Note
Coverage entities (i.e., Pilot-IDs) should be chosen carefully as calls routed to an unavailable entity are dropped.

For a user group, busy is defined as when all users in the user group are busy or in DND with at least one agent in the busy state. Coverage Pilot IDs will only be used if the user group's Backup Pilot IDs are unavailable or not defined.

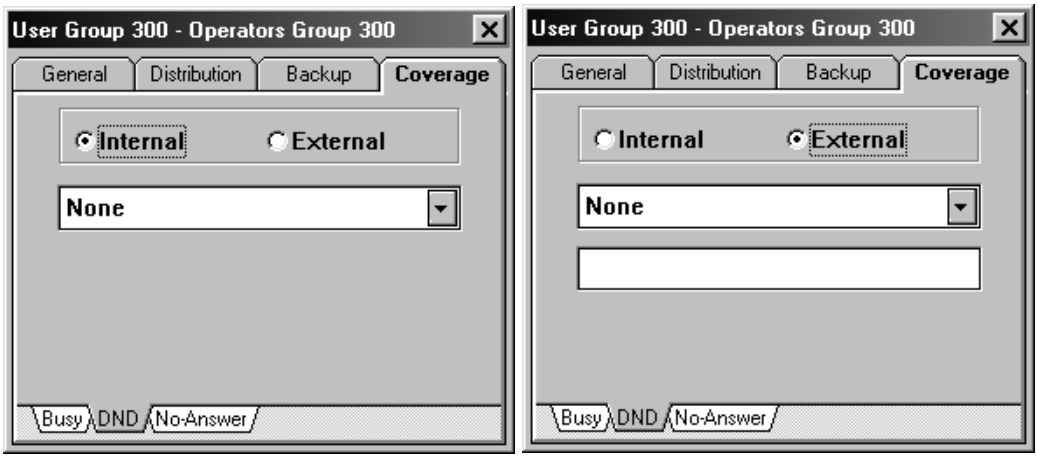
Note
In Coverage, if a Voice ID is set for the Pilot ID, the call will route to the user group's voice mailbox. For the call to route to the Auto Attendant, the user group's Backup must be set to the Voice ID.

7DEOH 8VHU *URXS &RYHUDJH %XV\

Pilot-ID on Busy: Internal	
Where the call will be transferred to when all users in the user group are busy or in DND (at least one agent must be in the busy state).	None * User ID User Group ID AutoAttendant ID Voice ID (see above note)
Pilot-ID on Busy: External	
Trunk or trunk group the call will be transferred to when the user group is busy. (This is used to route the calls to an external entity; e.g., answering service.)	None * Trunk ID Trunk Group ID
External Dial Number	
Externally dialed number for External Coverage.	none*, 0-9

* = default settings

8VHU *URXS &RYHUDJH '1'



Note
Coverage entities (i.e., Pilot-IDs) should be chosen carefully as calls routed to an unavailable entity are dropped.

For a user group, DND is defined as when all users in the user group are in DND. Coverage Pilot IDs will only be used if the user group's Backup Pilot IDs are unavailable or not defined.

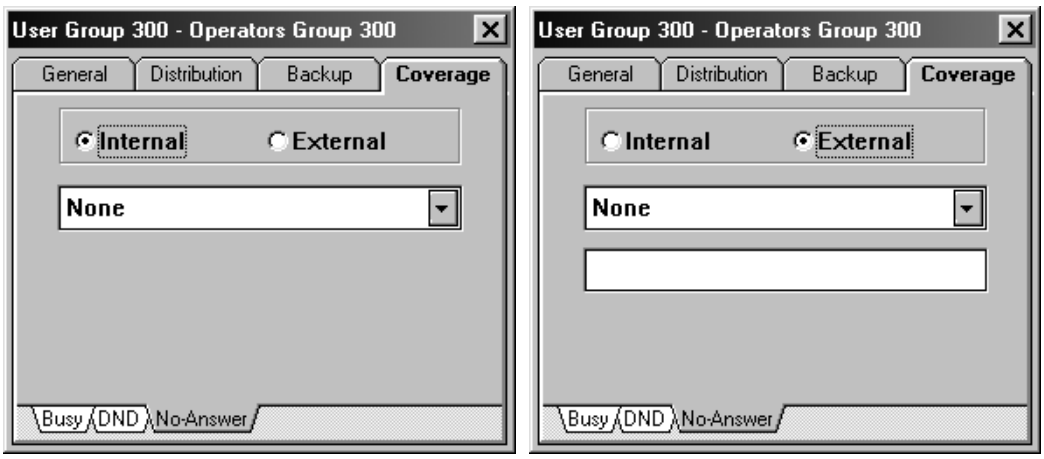
Note
In Coverage, if a Voice ID is set for the Pilot ID, the call will route to the user group's voice mailbox. For the call to route to the Auto Attendant, the user group's Backup must be set to the Voice ID.

7DEOH 8VHU *URXS &RYHUDJH '1'

Pilot-ID on Busy: Internal	
Where the call will be transferred to when all users in the user group are busy or in DND (at least one agent must be in the busy state).	None * User ID User Group ID AutoAttendant ID Voice ID (see above note)
Pilot-ID on Busy: External	
Trunk or trunk group the call will be transferred to when the user group is busy. (This is used to route the calls to an external entity; e.g., answering service.)	None * Trunk ID Trunk Group ID
External Dial Number	
Externally dialed number for External Coverage.	none*, 0-9

* = default settings

6.23.8 User Group: Coverage, No Answer



Note
Coverage entities (i.e., Pilot-IDs) should be chosen carefully as calls routed to an unavailable entity are dropped.

For a user group, No Answer is defined as when at least one user is not busy or in DND and the call is not answered after ringing to the available user(s). Coverage Pilot IDs will only be used if the user group's Backup Pilot IDs are unavailable or not defined.

Note
In Coverage, if a Voice ID is set for the Pilot ID, the call will route to the user group's voice mailbox. For the call to route to the Auto Attendant, the user group's Backup must be set to the Voice ID.

Table 6 - 72 User Group: Coverage, No Answer

Pilot-ID on Busy: Internal	
Where the call will be transferred to when there is no answer to the user group.	None * User ID User Group ID AutoAttendant ID Voice ID (see above note)
Pilot-ID on Busy: External	
Trunk or trunk group the call will be transferred to when there is no answer to the user group. (This is used to route the calls to an external entity; e.g., answering service.)	None * Trunk ID Trunk Group ID
External Dial Number	
Externally dialed number for External Coverage.	none*, 0-9

* = default settings

6.24 Configure trunks

1. Ensure that trunks have been created.
2. Click on the trunk tab to display the Trunk and Trunk Group dialog window.
3. Double-click on each trunk in the trunk window.
4. Make appropriate entries and selections on each of the tabs.

Tip
To alleviate the repetitive tasks associated with configuring multiple trunks, right-click on a setting or field and select **Replicate ALL** or **Replicate Selected** [Replicate Selected is only available if trunks have been selected]. The Replicate options automatically assign a setting to all or selected trunks.

6.24.1 Trunk: General

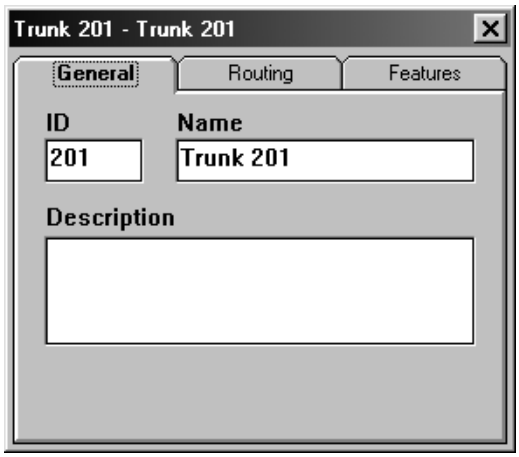


Table 6 - 73 Trunk: General

ID	
Logical identifier for the trunk. ***Note:*** The ID is automatically assigned based upon the Trunk ID Sequence entered when the first trunk was created.	Numeric Up to 5 digits
Name	
Informal identifier for the trunk. E.g., (512) 555-9500	Alphanumeric Up to 20 characters
Description	
Informal description or notes about the trunk. E.g., main CO line	Alphanumeric

6.24.2 Trunk: Routing: Voice

Note
Each Trunk ID in the Trunk dialog window will be highlighted in red until the routing for the Trunk has been configured.

Table 6 - 74 Trunk: Routing: Voice

Day Calls Routing Method	
Method by which the system will route calls during the Day Mode. ***Note:*** The Day Calls: Pilot should still be set if Fixed is not chosen. This will serve as the default routing in case other routing methods do not receive the required inputs from the local service provider (see “Trunk Routing” - section 8.133).	Fixed * DID Mapped-DID / DNIS Caller-ID / ANI
Day Calls: Pilot	
Where the system will route the incoming calls during the day hours [See “System parameters: Time, Mode” - section 6.21.6].	None * User ID User Group ID Auto Attendant ID Voice ID
Night Calls Routing Method	
Method by which the system will route calls during the Night Mode. ***Note:*** The Route Call during Day/Night should still be set if Fixed is not chosen. This will serve as the default routing in case other routing methods do not receive the required inputs from the local service provider (see “Trunk Routing” - section 8.133).	None * Trunk ID Trunk Group ID

Table 6 - 74 Trunk: Routing: (continued)Voice

Night Calls: Pilot	
Where the system will route the incoming calls during the night hours [See “System parameters: Time, Mode” - section 6.21.6].	None * User ID User Group ID Auto Attendant ID Voice ID

* = default settings

6.24.3 Trunk: Routing: Fax

Table 6 - 75 Trunk: Routing: Fax

Day Calls Pilot	
Where the system will route the incoming fax calls during the Day Mode [See “System parameters: Time, Mode” - section 6.21.6]. ***Note:*** The Plexus system must be properly configured to detect fax tones on incoming calls. See “Fax Detection” - section 8.73 for setup parameters.	None * User ID User Group ID AutoAttendant ID Voice ID
Night Calls Pilot	
Where the system will route the incoming fax calls during the Night Mode [See “System parameters: Time, Mode” - section 6.21.6]. ***Note:*** The Plexus system must be properly configured to detect fax tones on incoming calls. See “Fax Detection” - section 8.73 for setup parameters.	None * User ID User Group ID AutoAttendant ID Voice ID

* = default settings

6.24.4 Trunk: Features

Table 6 - 76 Trunk: Features

DISA Enabled		
Enables DISA and Remote Station Service on the trunk.	Enable/Disable *	
DISA Activation Timeout		
The period of time allotted for a caller to enter the DISA access digit. ***Note:*** If a DISA password is not entered, the call is routed according to the defined routing [See “Trunk: Routing: Voice” - section 6.24.2].	0-255 seconds	2 seconds*
DISA Session Timeout		
Period of time before a DISA or Remote Station Service session is automatically terminated. ***Note***: The session timeout is provided to limit the duration of potentially fraudulent DISA sessions and/or to control costs.	0-65,535 minutes	15 minutes *

* = default settings

6.25 Configure trunk groups

1. Ensure that trunks and trunk groups have been created.
2. Click on the trunk tab to display the Trunk and Trunk Group dialog window.
3. Double-click on the trunk group in the trunk group window.
4. Make appropriate entries on the tabs.

6.25.1 Trunk groups: General

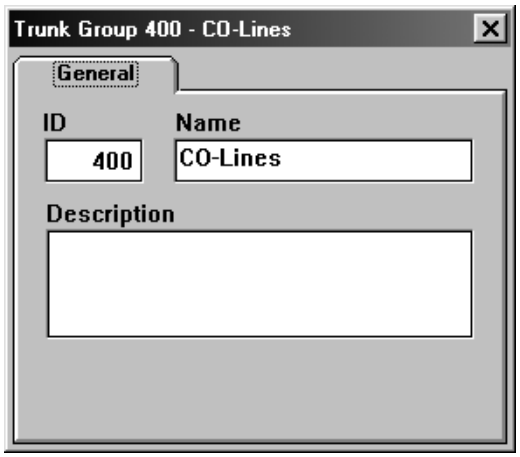


Table 6 - 77 Trunk groups: General

ID	
Logical identifier for the trunk group. ***Note:*** The ID is automatically assigned based on the Trunk Group ID sequence entered when the first trunk group was created.	Numeric Up to 5 digits
Name	
Informal identifier for the trunk group. E.g., Telemarketing	Alphanumeric Up to 20 characters
Description	
Informal description or notes about the trunk group. E.g., These trunks are used for outbound telemarketing	Alphanumeric

6.26 Saving a configuration

To save a completed configuration, proceed as follows:

1. From the **File** menu, select **Save As**.
2. Type the desired name on the **File name** line (up to 8 characters).
3. Select the desired directory.
4. Click on **OK**.

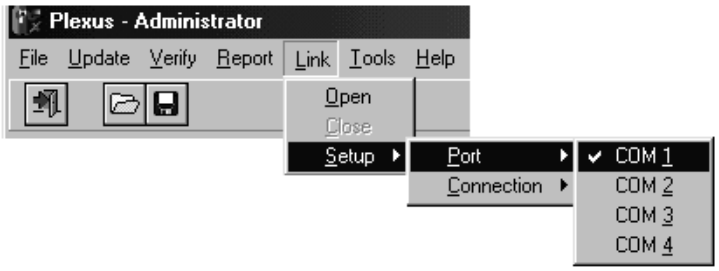
The file will be saved with a .zdb extension.

6.27 Establishing a link

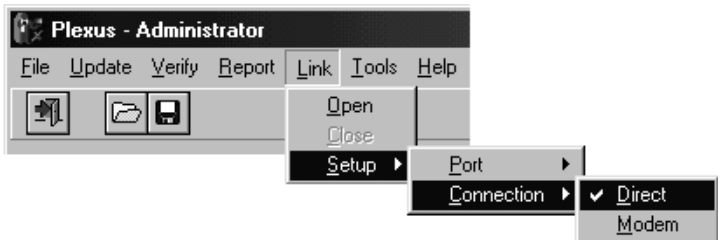
1 Connect the serial port on the DXP card (labeled “RS-232”) to an available COM port on a PC using a 9-pin serial cable.

2 From the **Link** menu, select **Setup** and **Port** (as shown below). Then, choose the COM port to which the serial cable is connected.

Note
The chosen COM port must be an *available* COM port [See “About serial ports and COM ports” - section 6.27.1].



3 From the **Link** menu, select **Setup** and **Connection** (as shown below). Then, choose **Direct.**



Note
The default Connection setting, Direct (**Connection** and **Direct**), applies to a serial connection. Modem connections apply only when performing Remote System Management.

6.27.1 About serial ports and COM ports

Most computers provide two serial ports for use by serial devices. Serial ports are either 9-pin or 25-pin external ports. Each serial port has an associated COM port. Microsoft Windows supports COM 1, COM 2, COM 3, and COM 4. Generally, the two serial ports are pre-configured as COM 1 and COM 2. Internal devices such as modems may utilize a COM port. Should an internal device utilize COM 1 or COM 2, the associated serial port will be unavailable. Should an internal device utilize COM 3 or COM 4, the serial ports are available as long as the COM port utilized by the internal device has been assigned other than the default IRQ.

Tip

Should the computer's only available serial port be a 25-pin port, a 9-25 pin adapter must be used. 9-pin serial cables and 9-25 pin adapters may be purchased at any electronics store.

6.27.2 About IRQs

An interrupt request (IRQ) is a unique number (between 0-15) assigned to each device in a computer. The unique number assigned to a device enables the computer's processor to manage which device is receiving its attention. If devices share an IRQ, a conflict arises as the processor does not know which device to attend to. COM 1 and COM 3 share a default IRQ. COM 2 and COM 4 share a default IRQ. In order to simultaneously use COM 1 and COM 3 or COM 2 and COM 4, one of the COM ports must be assigned a different IRQ (i.e., other than the default)

6.28 Uploading the configuration file

1. Save the desired configuration and keep the file open.
2. From the **Link** menu, select **Open**.
3. The link indicator at the bottom of the screen should indicate **Link: Opened** and the LED image should appear green. A magnifying glass will appear on the toolbar. To verify an active link, press the magnifying glass. This will query the current XOS version of the system. If the window showing the XOS version is displayed, you have a good link.
4. From the **Update** menu, select **Full**.

A status indicator at the bottom of the screen will display the progress of the update.

6.29 Updating the system clock

The Plexus system clock may be updated using Plexus Administrator as follows:

1. Launch Plexus Administrator.
2. Open a configuration file (.zdb).

Note
A configuration file must be open for the **Update** menu to be available. The open configuration file will not be uploaded or used in any way to update the clock.

3. From the **Link** menu, select **Open**.

The link indicator at the bottom of the screen should indicate Link: Opened and the LED image should appear green. A magnifying glass will appear on the toolbar. To verify an active link, press the magnifying glass. This will query the current XOS version of the system. If the window showing the XOS version is displayed, you have a good link.

4. From the **Update** menu, select **Clock**.

The Plexus system clock will be updated with the clock settings from the PC.

6.30 Updating Plexus key telephone programming

6.30.1 For Version 1.01 Key Telephones

These steps are for use with Key Telephones version 1.01 only.

The 20 programmable keys on a version 1.01 Plexus Key Telephone may be programmed using Plexus Administrator [Refer to "User: Features, Settings" - section 6.22.3]. After completing the key telephone programming (for all of the users on the system), and with the configuration file open:

1 From the **Link** menu, select **Open**.

The link indicator at the bottom of the screen should indicate Link: Opened and the LED image should appear green

2 From the **Update** menu, select **Key-Phone**.

Note
Updating Key Telephones through the **Update - Key Phone** option will cause each version 1.01 Key Telephone to reset, disconnecting active calls.

6.30.2 For Version 2.0 Key Telephones or later

These steps are for use with Key Telephones version 2.0 and later only.

The programmable keys on a version 2.0 and later Plexus Key Telephone may be programmed using Plexus Administrator [Refer to "User: Features, Settings" - section 6.22.3].

1 From the **Link** menu, select **Open**.

The link indicator at the bottom of the screen should indicate **Link: Opened** and the LED image should appear green.

2 Open the window User: Features: Settings: Define Key for the user's Key Telephone being programmed.

3 Complete the Key Telephone programming for the user.

4 Press **Update** button. This will update only this user's Key Telephone.

Note
The **Update now** button will not be an available option unless the window was opened while having an active link with the system.

A. or to program all Key Telephones

After completing the key telephone programming (for all of the users on the system), and with the configuration file open:

1 From the **Link** menu, select **Open**.

The link indicator at the bottom of the screen should indicate Link: Opened and the LED image should appear green

2 From the **Update** menu, select **Key-Phone**.

Note
Updating Key Telephones through the **Update - Key Phone** option will cause each version 1.01 Key Telephone to reset, disconnecting active calls.

6.31 Updating system XOS

Note
When the XOS is upgraded and the system is reset, the configuration will be deleted and will need to be updated before the system will work. This will not affect the voice mailboxes or the Auto Attendant structure.

1. Launch Plexus Administrator.
2. Open a Plexus configuration file.
3. From the **Link** menu, select **Open**.
4. Right mouse-click on the DXP card.
5. Select **Upgrade XOS**.
6. Select the directory and file name where XOS file is located.
7. Press **OK**.
8. Wait for the message **Plexus XOS Upgrade Successful** signaling that the upgrade is complete.
9. You will be prompted whether or not to **Reset the System now.** The XOS upgrade will not be complete until the system is reset.
10. Press **Yes** or **No**. If you press **No**, the system can be manually reset later.
11. After the system has reset, from the **Update** menu, select **Full** to update the configuration.

7

Key Telephone Software Configuration

7.1 Overview

Plexus system configurations may be minimally modified through a Plexus key telephone with an LCD display (i.e. PVT 30D or PDT 30D). Key telephone configuration is an easy way to make quick changes to the various system settings. Key telephone modification is most useful for maintenance activities on an established system. Maintenance activities include adding users to a user group, adding phone numbers to the system speed dial, or setting the system time.

Note
To proceed with key telephone configuration, the Plexus system must be powered up with all peripheral cards inserted [Refer to chapter 4, "Installation" for guidance on inserting peripheral cards].

7.2 Entering Programming mode

Programming mode may be activated on any Plexus key telephone (with a display) at any extension on the system.

To activate programming mode, proceed as follows:

1. Press **DND/FWD** to put the extension in DND mode.
2. Place the included Key Telephone Configuration template over the 10 keys at the bottom right-hand corner of the Plexus key telephone.

The Key Telephone Configuration template indicates the functions of the keys while in programming mode. If a template is not available, use the following programming equivalency table.

Programming	Key Telephone	Definition
Enter	SPKR	Select a menu
Save/Back	HOLD	Save changes and go back to the previous menu level
Previous	VOL ↓	Return to the previous menu or attribute
Next	VOL ↑	Proceed to the next menu or attribute
Prev Options	CONF	Return to the previous option
Next Options	XFER	Proceed to the next option
Backspace	DND/FWD	Backspace
Clear	MSG	Clear field

3. Press and hold **OVERRIDE** down for 2 seconds.
4. Press **PREVIOUS** or **NEXT** until **Setup PBX** appears on screen.
5. Press **ENTER**.

6 Enter the System Administrator password. The default password is **123456**.

7 Press **ENTER**.

Programming mode has now been activated.

8 Use **PREVIOUS** and **NEXT** to scroll through the programming menus.

Notes

On a Plexus PVT-22D key telephone, the **MUTE** button serves as the **OVERRIDE** button.

When a key telephone is in programming mode, it is available to receive both internal and external calls. It is recommended that you place the key telephone in Do Not Disturb mode before entering the programming mode.

7.3 Entering alphanumeric characters

While in programming mode, alphanumeric characters are entered according to the following table. These characters are used when entering names of users, user groups, or trunks.

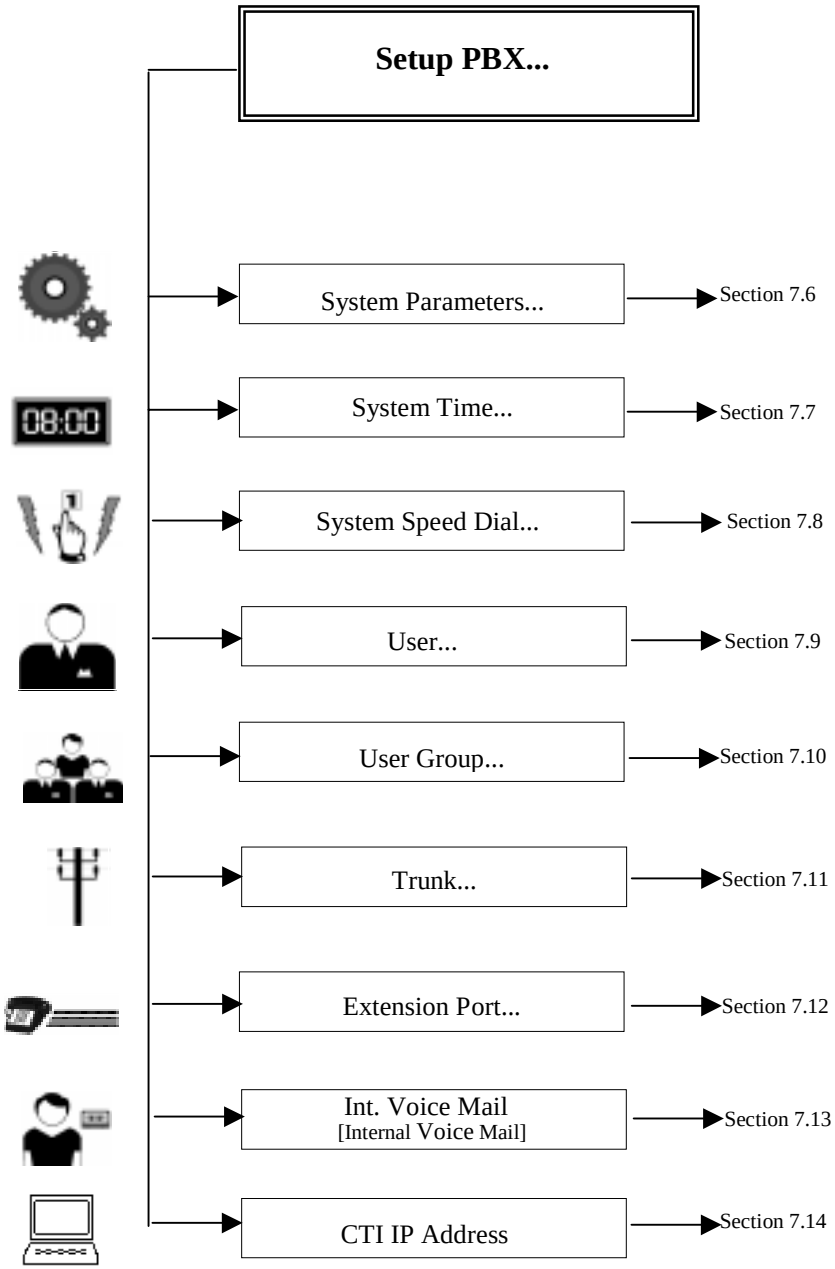
Table 7 - 1 Alphanumeric characters

Programming	Key Telephone	Programming	Key Telephone
0-9, *, #	Numeric keypad	O/o	CALL 3
A/a	17	P/p	05
B/b	18	Q/q	06
C/c	19	R/r	07
D/d	20	S/s	08
E/e	CALL 1	T/t	SCREEN ˆ
F/f	13	U/u	01
G/g	14	V/v	02
H/h	15	W/w	03
I/i	16	X/x	04
J/j	CALL 2	Y/y	CALLER ID
K/k	09	Z/z	FLASH
L/l	10	Space	REDIAL
M/m	11	UPPERCASE/ lowercase	CALL-BACK
N/n	12		

Note
Alphabetical characters may only be entered from a Plexus PVT-30D or PDT-30D key telephone. Apostrophes, as in the name **O'Reilly**, can only be entered through the Plexus Administrator.

7.4 Programming menus

The programming menus are as follows:

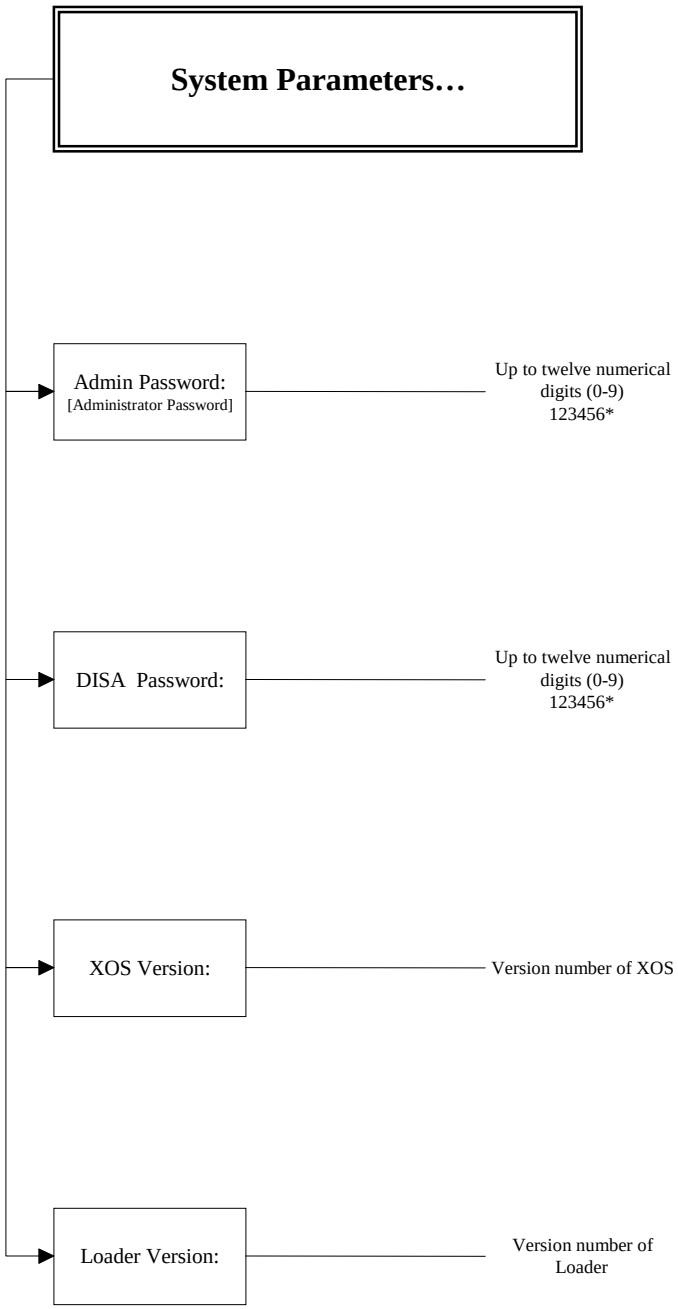


7.5 Exiting Programming mode

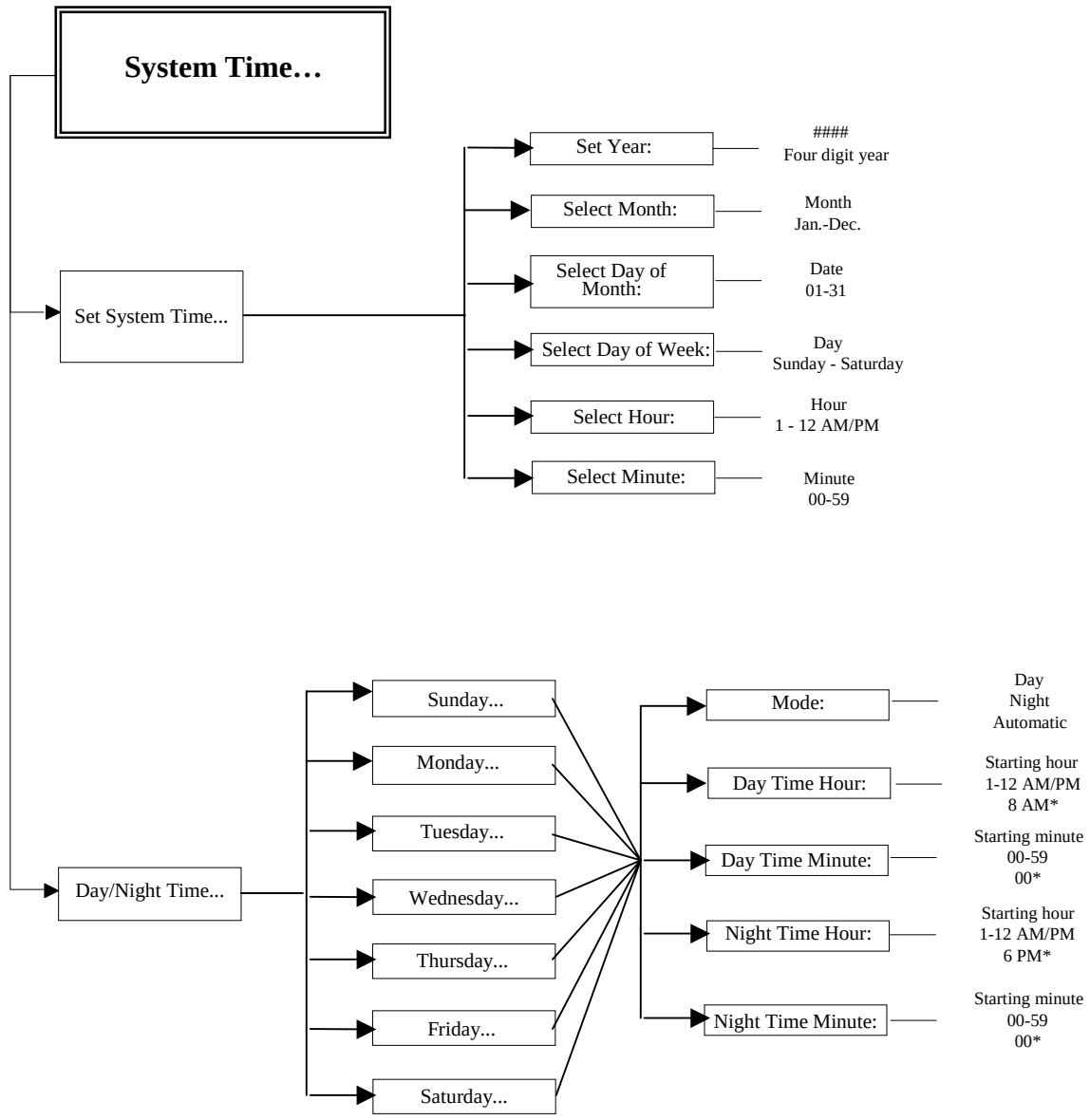
Upon completing a key telephone configuration session, the administrator must exit programming mode in order to make the key telephone available.

To exit programming mode, press and hold down **OVERRIDE** for 2 seconds.

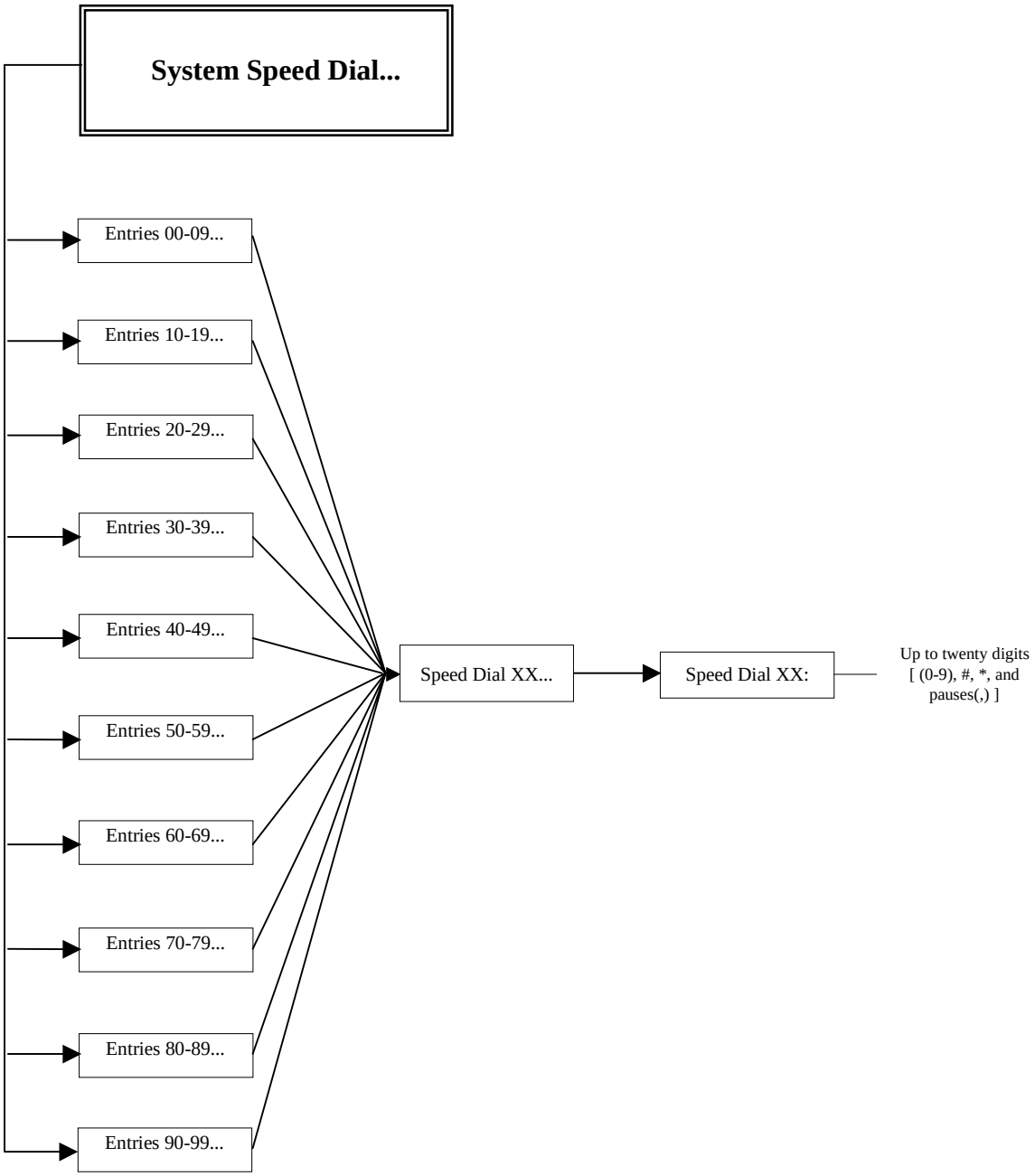
7.6 System Parameters



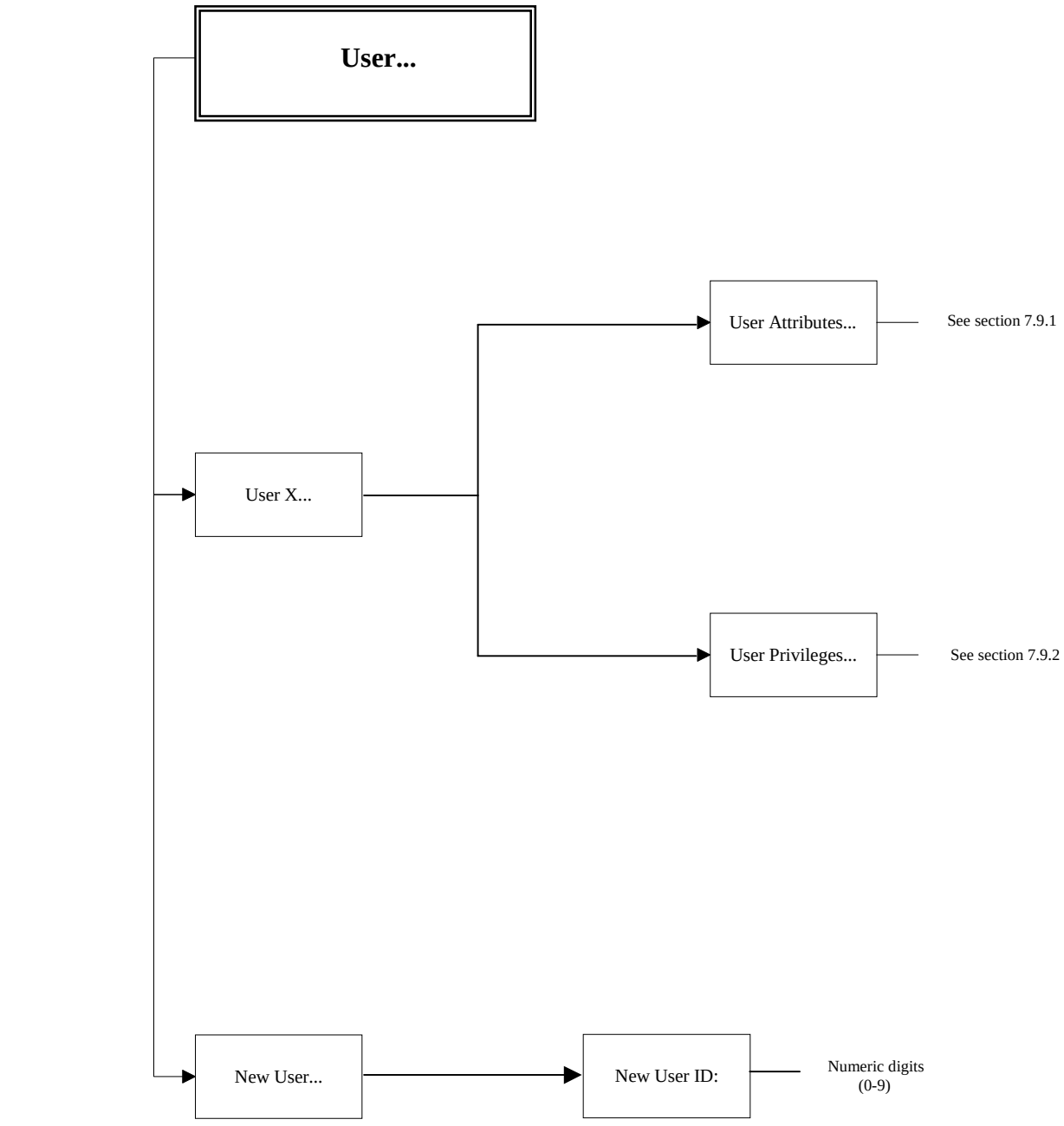
7.7 System Time



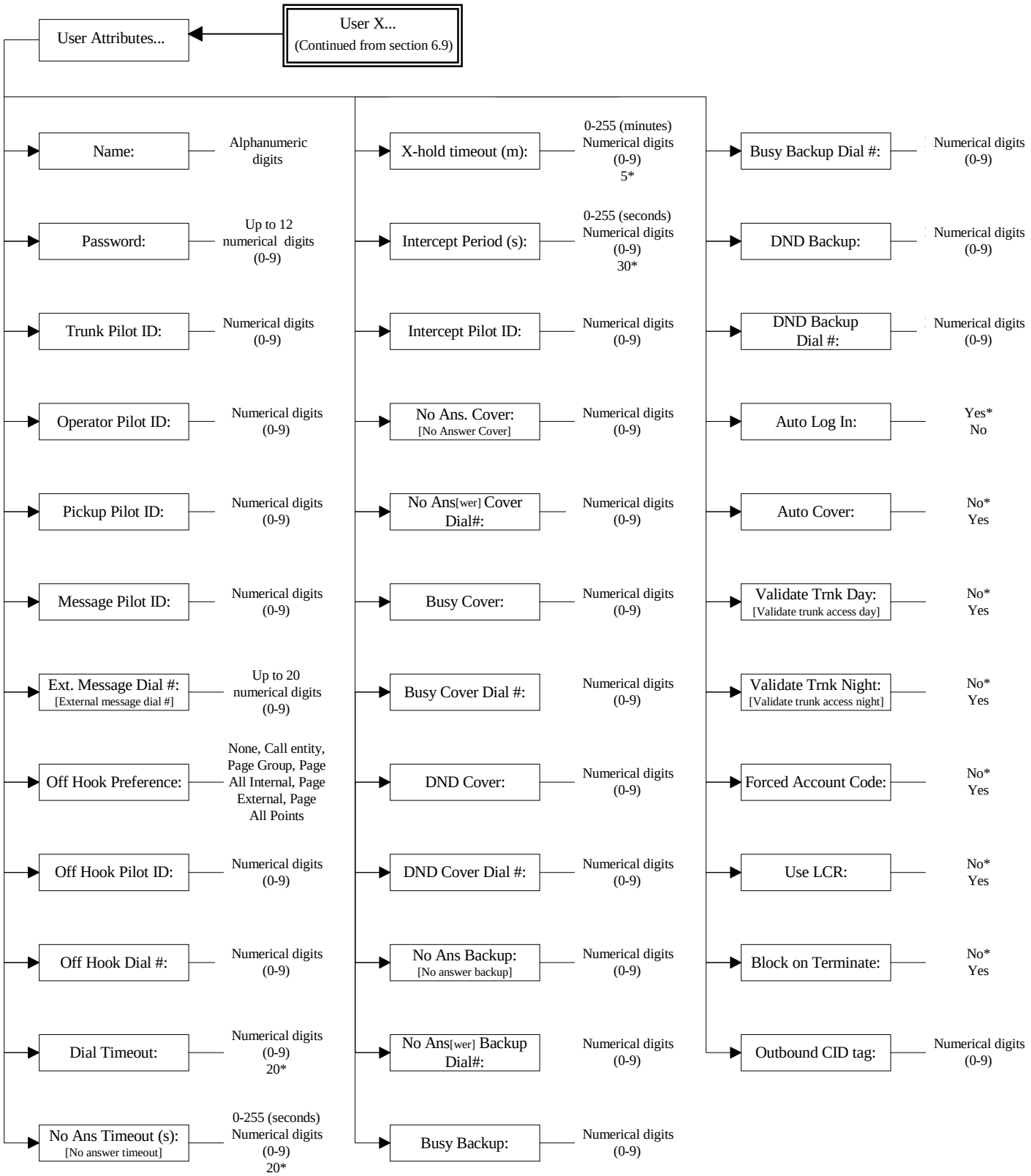
7.8 System Speed Dial



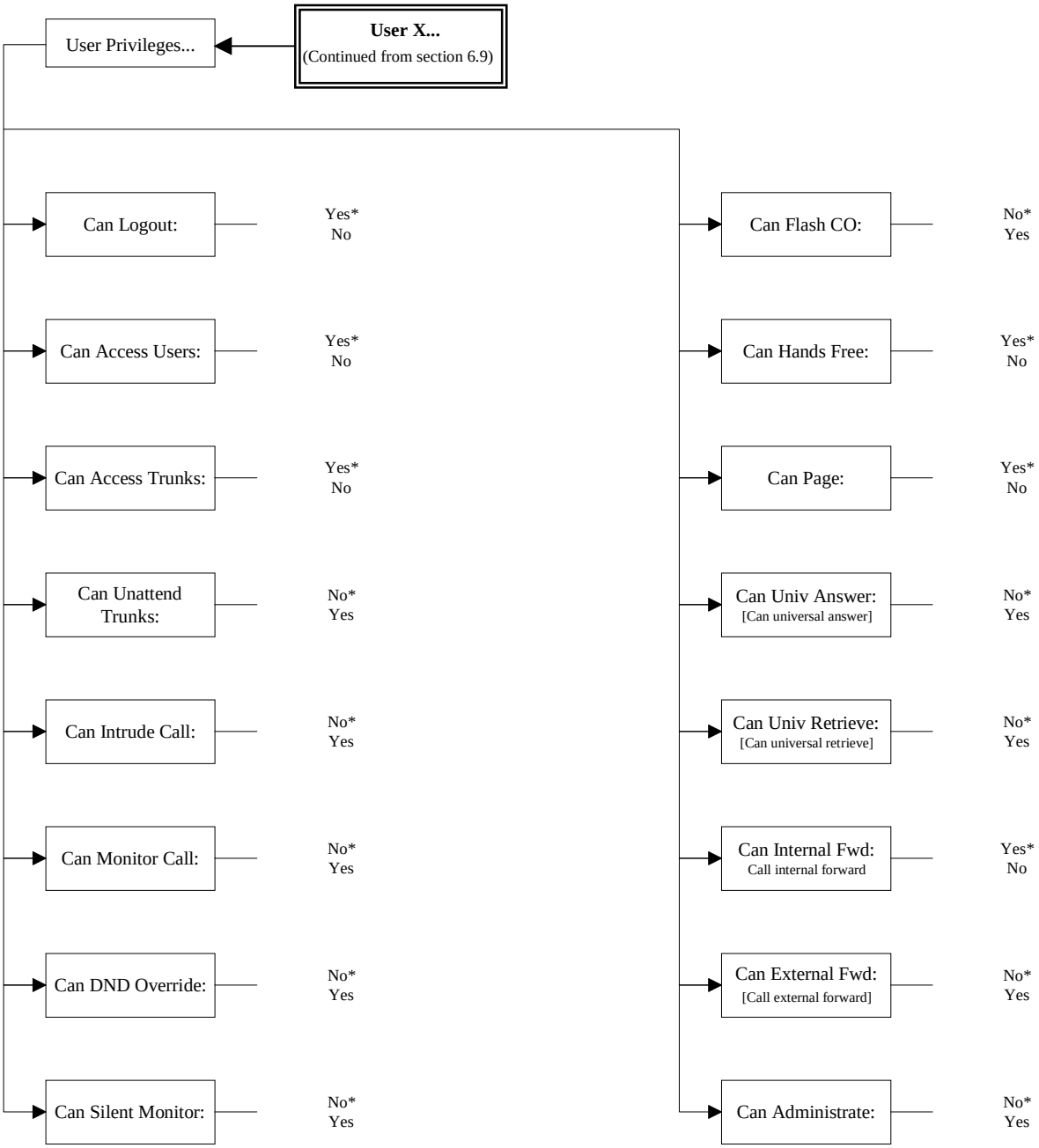
7.9 User


User...
User Attributes...
See section 7.9.1
User X...
User Privileges...
See section 7.9.2
New User...
New User ID:
Numeric digits
(0-9)

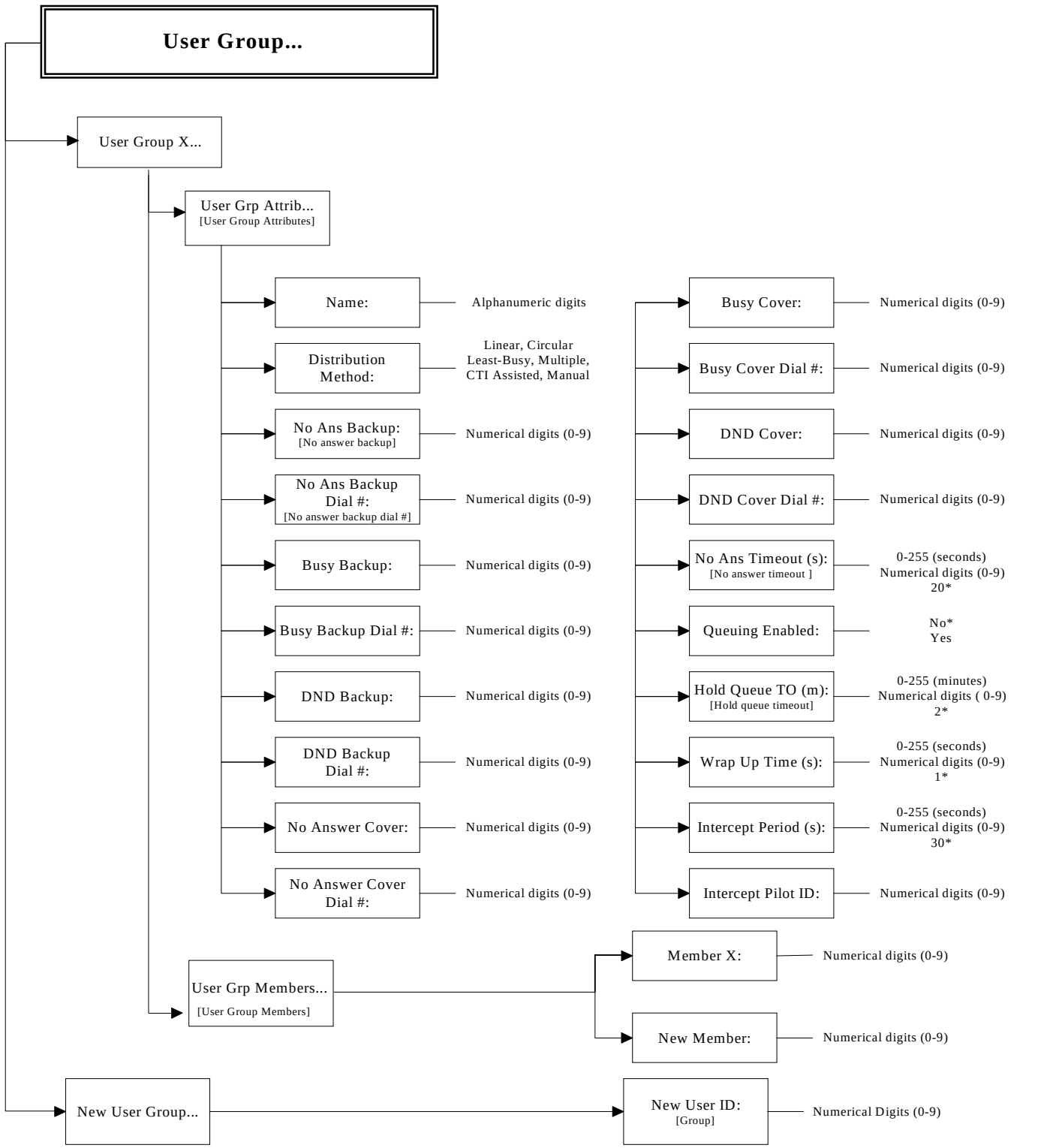
7.9.1 User Attributes



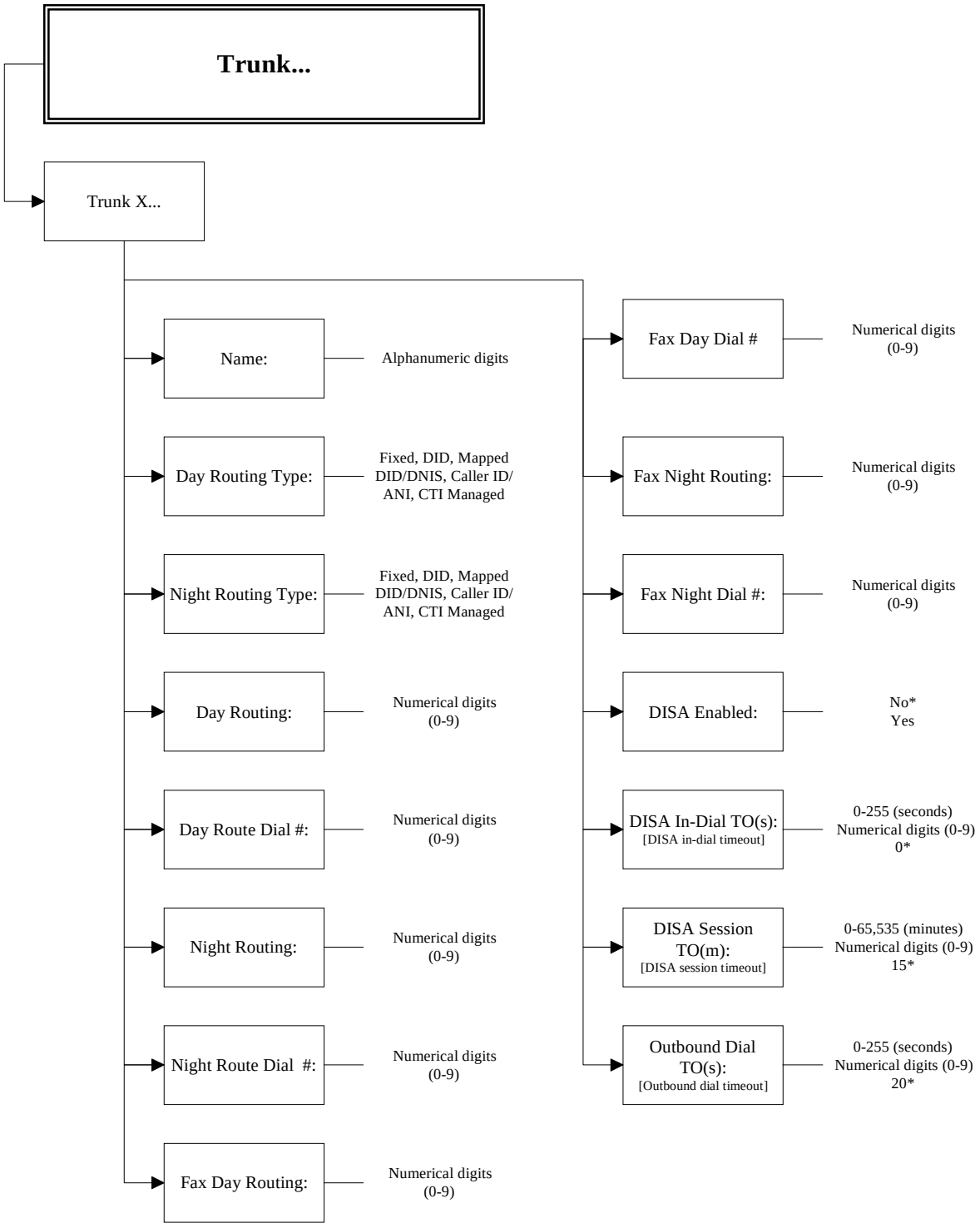
7.9.2 User Privileges



7.10 User Group

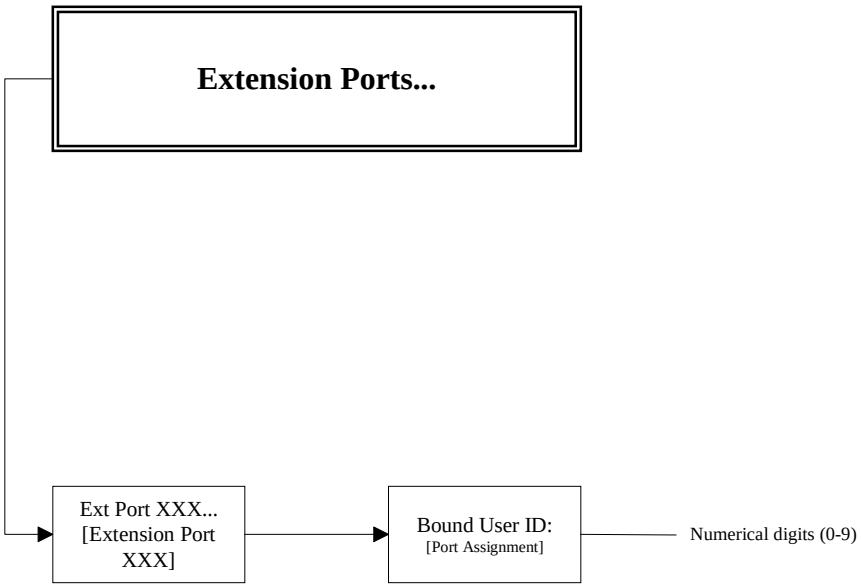


7.11 Trunk


Trunk...
Trunk X...
Name:
Alphanumeric digits
Day Routing Type:
Fixed, DID, Mapped DID/DNIS, Caller ID/ ANI, CTI Managed
Night Routing Type:
Fixed, DID, Mapped DID/DNIS, Caller ID/ ANI, CTI Managed
Day Routing:
Numerical digits (0-9)
Day Route Dial #:
Numerical digits (0-9)
Night Routing:
Numerical digits (0-9)
Night Route Dial #:
Numerical digits (0-9)
Fax Day Routing:
Numerical digits (0-9)
Fax Day Dial #
Numerical digits (0-9)
Fax Night Routing:
Numerical digits (0-9)
Fax Night Dial #:
Numerical digits (0-9)
DISA Enabled:
No*
Yes
DISA In-Dial TO(s):
[DISA in-dial timeout]
0-255 (seconds)
Numerical digits (0-9)
0*
DISA Session TO(m):
[DISA session timeout]
0-65,535 (minutes)
Numerical digits (0-9)
15*
Outbound Dial TO(s):
[Outbound dial timeout]
0-255 (seconds)
Numerical digits (0-9)
20*

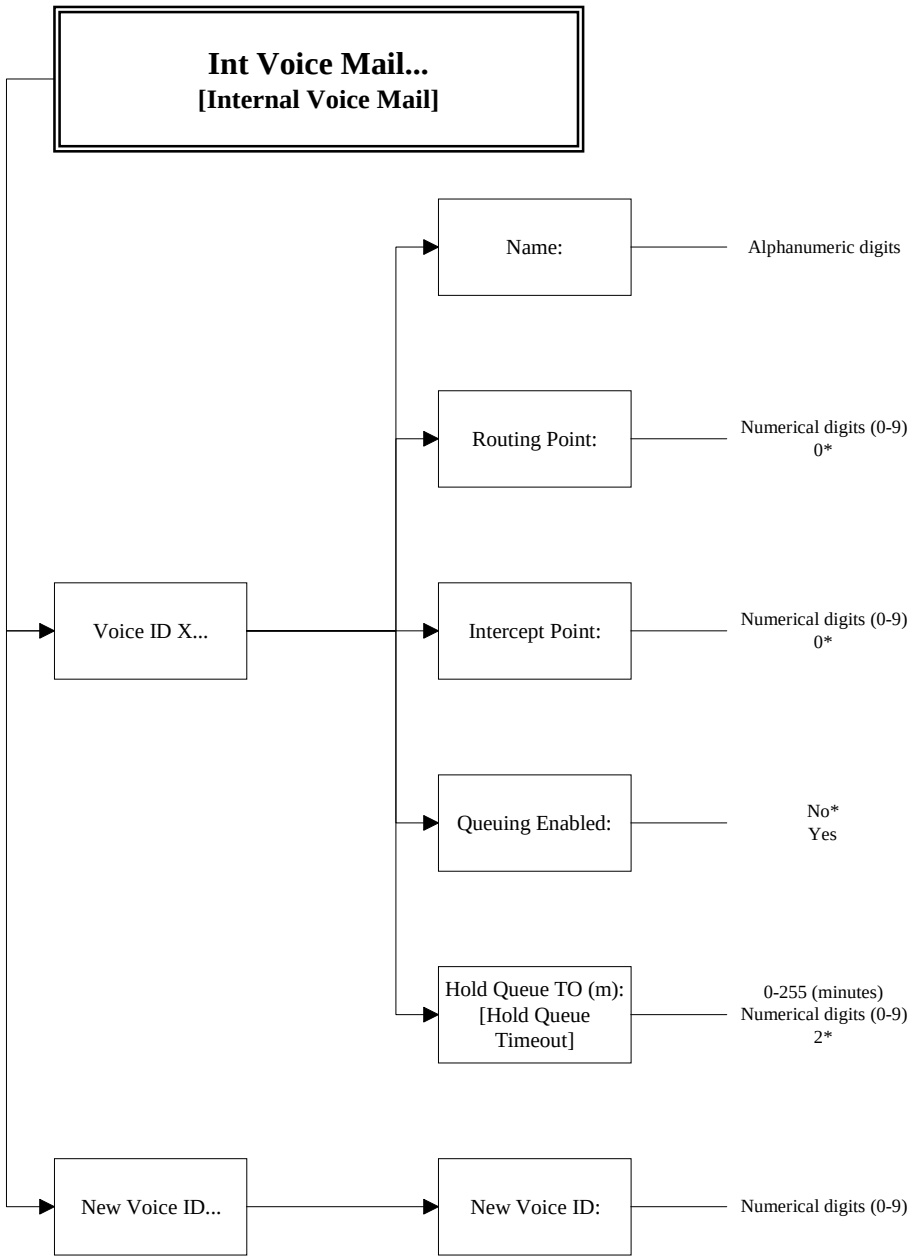
7.12 Extension Ports

Note
Key telephone configuration of extensions must be performed from an extension on the same unit as the extension port being configured. This applies when configuring Plexus systems linked with an optional Inter-Unit Link Interface (ILI) peripheral card.

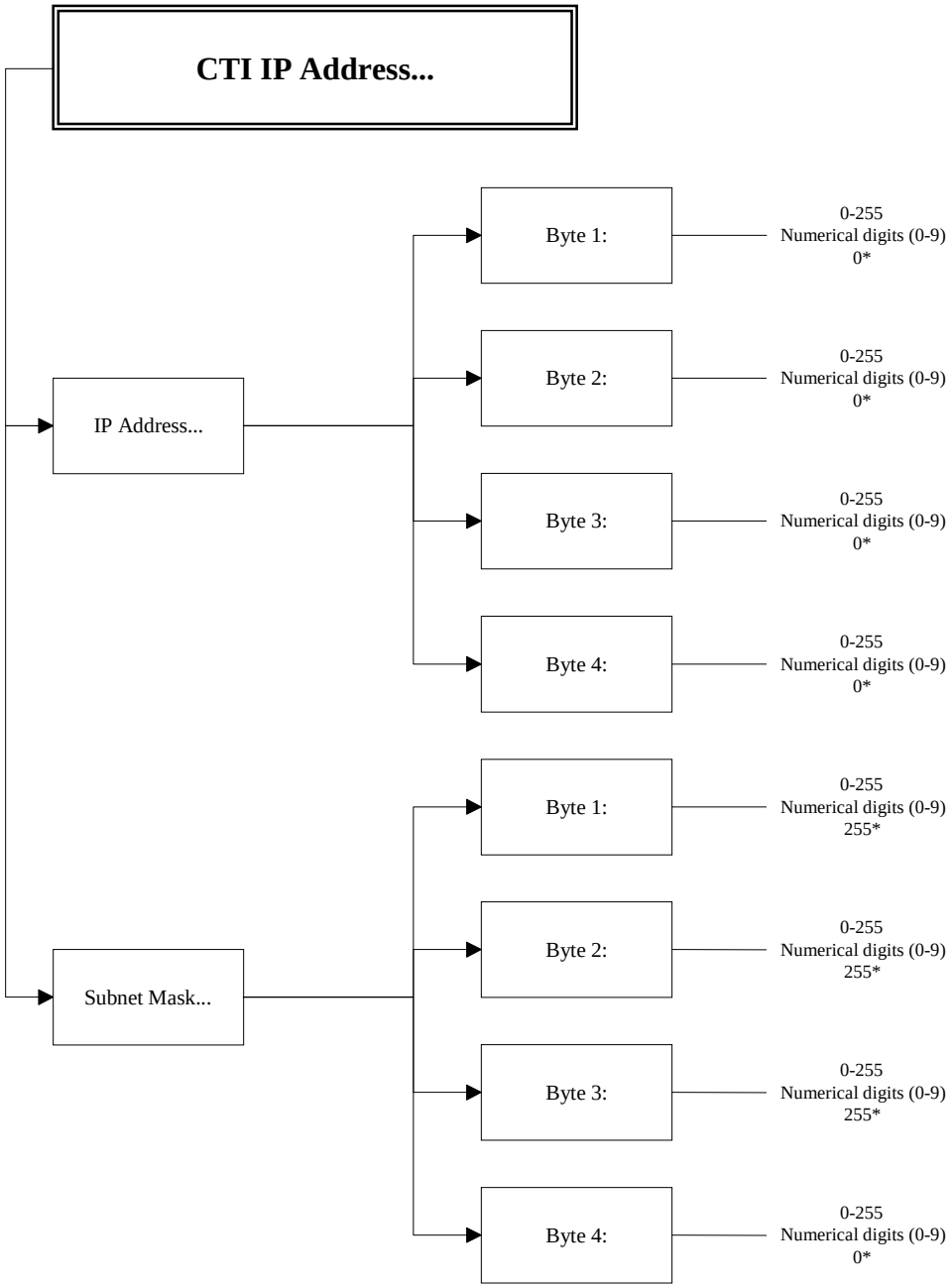


7.13 Int. Voice Mail

Note
These parameters are only present if an Integrated Voice Processor Interface has been installed in the system.



7.14 CTi IP Address



Feature Reference

8

This chapter provides information on the many features available with the Plexus system.

8.1 Overview of Features

This section provides an overview of the Features and Functions covered in the rest of this chapter. The following list describes the Feature Codes with a minimal description. The Feature Code list is followed by a Function Code list and by a Feature list of all Plexus system features covered in this chapter. The Plexus system features are listed alphabetically.

Note
In the chart below, **F** is a Feature Code which can be any dialed digit from 0 to 9 or * or #. Default = *. For more information on each Feature Code or Function Code in the charts below, refer to the section(s) listed under the Feature Section(s) heading of the chart.

8.1.1 Feature Code List

Table 8 - 1 Feature Code List

Feature Code	Required	Parameters	Feature Description	Action	Feature Section(s)
F00			**Call Hold : System** System Hold is a non-exclusive hold that enables held calls to be retrieved by any user on the system.	Places a call on Call Hold: System with your User ID/Holding Party's ID.	8.38
F01	Recipient User ID		**Call Hold: Park to Extension** Park is a non-exclusive system hold that enables held calls to be retrieved by any user on the system.	Places a call on System Hold with the Recipient's ID. Use F05 to retrieve.	8.38
F02			**Call Hold : Exclusive** Exclusive Hold enables a user to place calls on hold in such a way that no other user on the system can retrieve the call by pressing the associated DSS/DTS key.	Places a call on Call Hold: Exclusive	8.37
F03			**Call Retrieve : Exclusive or System** Call Retrieve is used to retrieve calls placed on either Exclusive Hold or System Hold.	Retrieves any type of held call to the responsible user (i.e., any call placed on Exclusive Hold).	8.50
F04	Trunk ID (or user ID if the held party is a system user)		**Call Retrieve : Exclusive or System** Call Retrieve is used to retrieve calls placed on either Exclusive Hold or System Hold.	Retrieves a specific call on System Hold by entering a feature code and the trunk ID of the held call (or user ID if the held party is a system user).	8.50
F05	Recipient User ID		**Call Retrieve : Exclusive or System** Call Retrieve is used to retrieve calls placed on System Hold with F01 - Park to extension. Retrieves calls on exclusive hold from another extension	Retrieves a tagged call on System Hold by entering the Parked to extension ID. Retrieves a call on exclusive hold by entering the extension the call was placed on exclusive hold from.	8.50
F06			**Call Retrieve : System** Call Retrieve is used to retrieve calls placed on either Exclusive Hold or System Hold.	Retrieves any call on System Hold according to a First In First Out (FIFO) convention.	8.50
F07			**Call Pickup : Intra-Group** Intra-Group Call Pickup enables a user to answer the ringing station of any user in their assigned pickup group.	Answers any ringing station within the pickup group.	8.44
F08	User ID		**Call Pickup : Intra-Group** Intra-Group Call Pickup enables a user to answer the ringing station of any user in their assigned pickup group.	Answers a specific ringing station within the pickup group.	8.44
F09			**Reserved**		
F10			**Do-Not-Disturb (DND) - Deactivate** Do Not Disturb (DND) allows a user to prevent all internal and external calls from ringing at their station. Paging announcements are also disabled.	Deactivate DND mode.	8.68
F11			**Do-Not-Disturb (DND) - Activate** Do Not Disturb (DND) allows a user to prevent all internal and external from ringing at their station. Paging announcements are also disabled.	Activates DND mode.	8.68

Table 8 - 1 Feature Code List

Feature Code	Required Parameters	Feature Description	Action	Feature Section(s)
F12		**DND Override** Do Not Disturb (DND) Override allows a user to override the DND mode of another user.	Override DND mode of another user.	8.69
F13		**Call Monitor** Call Monitor allows a user, with the appropriate privileges, to monitor or eavesdrop on a busy line or user.	Activates Call Monitor.	8.40
F14		**Whisper Page** Whisper Page allows a user to page a target user over the headset/speakerphone while the target user is on a call.	Pages busy user through handset/ speaker without interrupting the conversation.	8.141
F15		**Call Coach** Call Coach allows a user to coach a target user on a call without the other party being able to hear the coaching	Allows 2nd user to coach user during a call without the other party hearing the coaching.	8.29
F16		**Call Cover** When a user receives no answer, a busy signal, or a Do Not Disturb (DND) indication on an internal call, the Call Cover feature may be utilized to reach the unavailable user's call coverage (e.g., voice mailbox, operator).	Routes user to target user's coverage	8.31
F17		**Camp-On** Camp-On enables a user to place themselves into the busy hold queue of the target user or user group.	Places the call in the Hold Queue of the target user.	8.58
F18		**Call Back** The Call Back feature allows users to request a call back from the system when an unavailable entity becomes available.	Initiate Call Back	8.25 8.26
F19		**Call Back - Cancel** Cancels Call Back	Cancel call backs.	8.25 8.26
F20		**Auto-Answer - Deactivate** The Auto-Answer feature allows users with Plexus Key Telephones to automatically answer all internal calls in hands-free mode.	Deactivates Auto-Answer.	8.13
F21		**Auto-Answer - Activate** The Auto-Answer feature allows users with Plexus Key Telephones to automatically answer all internal calls in hands-free mode.	Activates Auto-Answer.	8.13
F22	User ID	**Hands-Free Call** Hands-Free Call allows a Plexus Key Telephone user with the appropriate privileges to place hands-free calls to other Plexus Key Telephone users. A Hands Free Call causes the target user's key telephone to automatically go off-hook in speakerphone mode.	Initiates a hands-free call.	8.79
F23		**Page – Internal, All Users** The Internal Paging facility allows users to page through Plexus Key Telephone speakers.	Pages over *all* key telephones.	8.105
F24	User Group ID	**Page – Internal, Specific User Group** The Internal Paging facility allows users to page through Plexus Key Telephone speakers.	Pages over a specific group of key telephones.	8.105
F25		**Page – External** The External Paging facility allows users to page through a customer-provided external (e.g., overhead) paging system.	Places a page over an external facility.	8.104

Table 8 - 1 Feature Code List

Feature Code	Required	Parameters	Feature Description	Action	Feature Section(s)
F26			**Page – All Points** The All Points Paging facility allows users to page through both the Plexus Key Telephones and a customer-provided external (e.g., overhead) paging system.	Places an all points page.	8.101
F27			**Page – Answer** Answer Page allows a user to respond to a page by being automatically connected to the paging party.	Answers a page.	8.102
F28			**Page – Block - Deactivate** Page Block allows a user to block the broadcast of pages through their Plexus Key Telephone.	Cancels/deactivates Page Block.	8.103
F29			**Page – Block - Activate** Page Block allows a user to block the broadcast of pages through their Plexus Key Telephone.	Blocks paging	8.103
F30			**Log Out (User Authorization)** Users log in or sign in to the system with their user ID. When a user logs in or signs in, the system knows where they are located and routes calls accordingly.	Logs out of the system.	8.11 8.116
F31	User ID	Password	**Log In (User Authorization)** Users log in or sign in to the system with their user ID. When a user logs in or signs in, the system knows where they are located and routes calls accordingly.	Logs into the system.	8.11 8.116
F32	User Group ID	Password	**Agent Log Off** Agent Log Off allows a user to log out of a user group	Logs User out of the User Group	8.5
F33	User Group ID	Password	**Agent Log On** Agent Log Off allows a user to log in to a user group	Logs User in to the User Group	8.5
F34	User Group ID		**Agent Not Ready** Agent Not Ready allows a user to not receive user group calls without using Log Off or going busy/DND	Places user in a busy status within the defined user group	8.6
F35	User Group ID		**Agent Ready** Agent Ready allows the user to receive user group calls after going Agent Not Ready.	Places user in an available status within the defined user group	8.6
F36			**Hands-Free Block - Deactivate** Hands-Free Block allows a user to block Hands-Free Calls from other users.	Cancels Hands-Free Block (unblock).	8.80
F37			**Hands-Free Block - Activate** Hands-Free Block allows a user to block Hands-Free Calls from other users.	Blocks calls from Hands-Free users.	8.80
F38			**Appointment Reminder - Cancel** The Appointment Reminder feature allows a user to program the system to ring them at a specified time as an appointment reminder.	Deactivates Appointment Reminder.	8.10
F39	Time (24 hour format)		**Appointment Reminder - Set** The Appointment Reminder feature allows a user to program the system to ring them at a specified time as an appointment reminder.	Activates Appointment Reminder.	8.10
F40	User ID	Password	**User Sign-Out** Users log in or sign in to the system with their user ID. When a user logs in or signs in, the system knows where they are located and routes calls accordingly.	Signs out on the system.	8.11

Table 8 - 1 Feature Code List

Feature Code	Required	Parameters	Feature Description	Action	Feature Section(s)
F41	User ID	Password	**User Sign-In** Users log in or sign in to the system with their user ID. When a user logs in or signs in, the system knows where they are located and routes calls accordingly.	Signs into the system.	8.11
F42	User ID	Password	**Authorize User** Authorize user allows a user to access their system privileges on another user's phone.	Temporarily transfers user's privileges to the extension.	8.12
F43			**Universal Answer: Any** Universal Answer: enables a user, with the appropriate privileges, to answer any ringing trunk, user, or user group.	Answers an unspecified ringing station:	8.134
F44	User ID / Trunk ID		**Universal Answer: Specific** Universal Answer: Specific enables a user, with the appropriate privileges, to answer any ringing trunk or station.	Answers a specific ringing station	8.134
F45	User ID		**Alarm Clock - Cancel** (*System Administrator Privilege)* The Alarm Clock feature allows users (e.g., front-desk personnel) to program the system to ring other users at a specified time as an alarm, notification, or wake-up call.	Deactivates alarm clock.	8.7
F46	User ID	Time	**Alarm Clock - Set** (*System Administrator Privilege)* The Alarm Clock feature allows users (e.g., front-desk personnel) to program the system to ring other users at a specified time as an alarm, notification, or wake-up call.	Activates alarm clock.	8.7
F47			**Day/Night Service - Return to Actual Time** *(System Administrator Privilege)* Day/Night Service enables Plexus systems to route incoming calls differently based on the day (Weekday or Weekend) and time (Day or Night) a call is received.	Returns system to normal settings (i.e. restores system to the mode programmed for that time of day).	8.62
F48			**Day/Night Service - Forced Day Mode** *(System Administrator Privilege)* Day/Night Service enables Plexus systems to route incoming calls differently based on the day (Weekday or Weekend) and time (Day or Night) a call is received.	Manually switches system to Day-Mode.	8.62
F49			**Day/Night Service - Forced Night Mode** *(System Administrator Privilege)* Day/Night Service enables Plexus systems to route incoming calls differently based on the day (Weekday or Weekend) and time (Day or Night) a call is received.	Manually switches system to Night Mode.	8.62
F50	User ID		**Message Waiting Indicator - Deactivate** (Hotel/Motel Features) Plexus systems are capable of providing Message Waiting Indication to Key Telephones and Ordinary Phones.	Manually deactivates Message Waiting Indicator.	8.87
F51	User ID		**Message Waiting Indicator - Activate** (Hotel/Motel Features) Plexus systems are capable of providing Message Waiting Indication to Key Telephones and Ordinary Phones.	Manually activates Message Waiting Indicator.	8.87
F52			**Access Own Personal Voice Mailbox** Allows users to check messages	Accesses a user's personal mailbox.	8.87 Chapter 9
F53	User ID		**Access Other Personal Voice Mailbox** Allows users to leave a message in another user's mailbox	Accesses the mailbox specified by the user.	Chapter 9

Table 8 - 1 Feature Code List

Feature Code	Required	Parameters	Feature Description	Action	Feature Section(s)
F54			**Flash – Central Office** Centrex features are often accessed by sending a 'flash' signal to the telephone company central office (CO). Because Plexus systems are capable of sending 'flash' signals to the CO, they are compatible with most Centrex services.	Sends 'flash' signals to the CO.	8.59
F55	Code		**Speed Dial: System** Speed dial allows users to utilize a system speed dial number by entering a feature code followed by the unique two-digit speed dial code.	Dials numbers stored in the system.	8.122
F56			**Redial: Last-Number** Last Number Redial enables a user to instruct the system to redial up to 20 digits of the last number the user dialed.	Redials last number dialed.	8.113
F57			**Reserved**		
F58			**Reserved**		
F59			**Reserved**		
F60			**External Call Queuing: User - Deactivate** Deactivates the user's external call queuing	Deactivates user's external call queuing	8.21
F61			**External Call Queuing: User - Activate** Activates the user's External Call Queuing	Activates user's External Call Queuing	8.21
F62	DN	#	**Call Forwarding: Immediate** Call Forwarding: Immediate enables users to immediately forward all their calls to an internal or external number.	Immediately forwards each call to the specified number.	8.35
F63	DN	#	**Call Forward on No Answer** Call Forward on No Answer enables users to have their calls forwarded to another user on the system when the caller receives no answer.	Forwards each call when the caller receives no answer.	8.35
F64	DN	#	**Call Forward on Busy** Call Forward on Busy enables users to have their calls forwarded to another user on the system when the caller receives a busy signal.	Forwards each call when the caller receives a busy signal.	8.35
F65	DN	#	**Call Forward on DND** Call Forward on DND enables users to have their calls forwarded to another user on the system when the caller receives a Do Not Disturb signal.	Forwards each call when the caller receives a DND signal.	8.35
F66	User ID		**Call Follow-Me - Deactivate** Cancels Call Follow Me	Cancels Call Follow-Me	8.34
F67	User ID		**Call Follow-Me - Activate** Call Follow-Me is a form of Internal Call Forwarding which can be activated by a user from a station other than the station at which they are logged in. When activated, all of the user's internal and external calls are immediately forwarded to their current location.	Immediately forwards all of the user's internal and external calls to their current location.	8.34
F68			**Reserved**		
F69			**Reserved**		
F70			**Call Waiting - Deactivate** The Call Waiting feature, when activated, provides a stutter tone to busy users to indicate that a call is waiting. The busy user can choose to answer the waiting call or allow it to route to No Answer coverage.	Deactivates Call Waiting.	8.54

Table 8 - 1 Feature Code List

Feature Code	Required	Parameters	Feature Description	Action	Feature Section(s)
F71			**Call Waiting - Activate** The Call Waiting feature, when activated, provides a stutter tone to busy users to indicate that a call is waiting. . The busy user can choose to answer the waiting call or allow it to route to No Answer coverage.	Activates Call Waiting.	8.54
F72	User ID		**Do-Not-Disturb - Deactivate** (System Administrator Privilege, Hotel/Motel Features) Places the target user's extension in Do not Disturb Mode.	Deactivates Do Not Disturb mode.	Chapter 18
F73	User ID		**Do-Not-Disturb - Activate** (System Administrator Privilege, Hotel/Motel Features) Places the target user's extension in Do not Disturb Mode.	Activates Do Not Disturb mode.	Chapter 18
F74	User ID		**Room Check Out** (System Administrator Privilege, Hotel/Motel Features) The Room Check Out feature may be activated by hotel personnel to disable a guestroom phone and designate the guestroom as 'Not Occupied, Not Serviced'.	Activates Room Check Out feature.	Chapter 18
F75	User ID		**Room Check In** (System Administrator Privilege, Hotel/Motel Features) The Room Check In feature may be activated by hotel personnel to reset a guestroom phone and designate the guestroom as 'Occupied'.	Activates Room Check In features.	Chapter 18
F76	User ID		**Room Occupancy Indicator - Deactivate** (System Administrator Privilege, Hotel/Motel Features) The Room Occupancy Indicator may be activated by hotel personnel to manually activate or deactivate the occupancy indicators.	Manually deactivates room occupancy indicator.	Chapter 18
F77	User ID		**Room Occupancy Indicator - Activate** (System Administrator Privilege, Hotel/Motel Features) The Room Occupancy Indicator may be activated by hotel personnel to manually activate or deactivate the occupancy indicators.	Manually activates room occupancy indicator.	Chapter 18
F78	User ID		**Room Service Indicator - Deactivate** (System Administrator Privilege, Hotel/Motel Features) The Room Service Indicator may be activated by hotel personnel to manually activate or deactivate the service indicators.	Manually deactivates room service indicator.	Chapter 18
F79	User ID		**Room Service Indicator - Activate** (System Administrator Privilege, Hotel/Motel Features) The Room Service Indicator may be activated by hotel personnel to manually activate or deactivate the service indicators.	Manually activates room service indicator.	Chapter 18
F80			**Transfer Recall - Deactivate** Cancels Transfer Recall	Deactivates Transfer Recall	8.131
F81			**Transfer Recall - Activate** Transfer Recall causes Blind Transferred calls that go unanswered at the target user/user group to ring back to the user that transferred the call rather than forwarding to the target user/user group's backup/coverage.	Activates Transfer Recall	8.131
F82	User ID		**Trunk Access - Disable** (System Administrator Privilege, Hotel/Motel Features) Trunk access codes are available to manually enable or disable access to outside phone lines from the guest room.	Manually deactivates access to outside trunks.	8.2 Chapter 18

Table 8 - 1 Feature Code List

Feature Code	Required	Parameters	Feature Description	Action	Feature Section(s)
F83	User ID		**Trunk Access - Enable** (System Administrator Privilege, Hotel/Motel Features) *Trunk access codes are available to manually enable or disable access to outside phone lines from the guest room.*	Manually activates access to outside trunks.	8.2 Chapter 18
F84	User ID		**User to User Access/Room to Room Access - Disable** (System Administrator Privilege, Hotel/Motel Features) Room to Room Access feature codes are available to manually enable or disable access to other rooms.	Manually deactivates access to other rooms.	8.2 Chapter 18
F85	User ID		**User to User Access/Room to Room Access - Enable** (System Administrator Privilege, Hotel/Motel Features) Room to Room Access feature codes are available to manually enable or disable access to other rooms.	Manually activates access to other users/ rooms.	8.2 Chapter 18
F86	User ID		**Failed Wake-Up Indicator - Deactivate** (System Administrator Privilege, Hotel/Motel Features) The System Administrator can also define a station (e.g., the Front Desk phone) to serve as the No-Response Cover Pilot. The No-Response Cover Pilot receives a failed alarm call when all retries of a wake-up call fail to reach the guest.	Deactivates Failed Wake-Up Indicator.	Chapter 18
F87	User ID		**Failed Wake-Up Indicator - Activate** (System Administrator Privilege, Hotel/Motel Features) The System Administrator can also define a station (e.g., the Front Desk phone) to serve as the No-Response Cover Pilot. The No-Response Cover Pilot receives a failed alarm call when all retries of a wake-up call fail to reach the guest.	Activates Failed Wake-Up Indicator.	Chapter 18
F88	Account #		**Account Code Entry** Account Codes offer users a way to tag calls with a numeric label of up to 12 digits in length. These codes are useful for billing and reporting purposes.	Assigns the entered account code to the call currently on hold.	8.3 8.4
F89			**Unsupervised Conference** The Unsupervised Conference feature allows a user, with the appropriate privileges, to release a call conference made up exclusively of external calls. Releasing such a conference allows the external calling parties to continue their call without a system user present.	Releases external parties from the call conference.	8.30 8.136
F90			**System Administrator Privilege - Disable** The maintenance and configuration of a number of features can only be performed from a station at which System Administrator Privileges (SAP) have been activated.	Deactivates System Administrator Privilege.	8.126
F91	Password		**System Administrator Privilege - Enable** The maintenance and configuration of a number of features can only be performed from a station at which System Administrator Privileges (SAP) have been activated.	Activates System Administrator Privilege.	8.126
F92			**Reserved**		
F93			**Reserved**		
F94			**Reserved**		

Table 8 - 1 Feature Code List

Feature Code	Required	Parameters	Feature Description	Action	Feature Section(s)
F95	Voice ID	Greeting Number	**Select Greeting** Selects the Route Point Starting Point in the Auto Attendant Node structure	Selects greeting in Auto Attendant	Chapter 09
F96	Auto Attendant ID		**Record Auto-Attendant** (System Administrator Privilege, Requires Auto-Attendant Card) The Record Auto-Attendant feature allows the administrator to Record a greeting for an AAI Card.	Records greeting for AAI card.	8.14 Chapter 15
F97	Port ID		**Enable Port** (*System Administrator Privilege)* Extension and trunk ports can be enabled or disabled.	Enables Port	8.108
F98	Port ID		**Disable Port** (*System Administrator Privilege)* Extension and trunk ports can be enabled or disabled.	Disables Port	8.108
F99			**Reset Extension Port Terminal** Resets the extension port and re-initializes Key Telephones.	Reset port and Key Telephone	8.8

8.1.2 Function Code List

Note
Function Codes are only available when programmed as a key on a Plexus key telephone. Functions are not available for ordinary phones.

Table 8 - 2 Function Code List

Function Code	Function Description	Action	Feature Section
Function 01	Call Record	Records current call to voice mail.	8.48
Function 02	Answering Machine Emulation / Live Call Screen	Enables a user to listen to callers as they leave a message in the user's voice mailbox	8.9
Function 03	Auto Answer	Enables users to have their phones answer internal calls automatically	8.13
Function 04	Day / Night Service	Shows the Time Mode of the Plexus system. Allows the user to switch time modes of the Plexus system	8.62
Function 05	Busy Queuing - User / External Call Queuing	Places external calls to the user into a hold queue when the user is busy	8.21
Function 06	Transfer Recall	Recalls unanswered calls that were transferred using Blind-Transfer	8.131

8.1.3 Feature List

The following list of Plexus system features is provided to simplify finding specific features in this chapter. Refer to the section number listed for feature descriptions.

Feature	Section
Overview of Features (Feature Code and Function Code lists)	8.1
Access Restriction: Line/User	8.2
Account Codes: Voluntary	8.3
Account Codes: Forced	8.4
Agent Log Off / Agent Log On	8.5
Agent Not Ready / Agent Ready	8.6
Alarm Clock	8.7
Analog Telephone/Device Support	8.8
Answering Machine Emulation	8.9
Appointment Reminder	8.10
Authorization Management	8.11
Authorize User	8.12
Auto-Answer	8.13
Automated-Attendant	8.14
Auto Camp-On	8.22
Auto Cover	8.15
Auto Voltage Selection	8.16
Automatic Line Selection	8.17
Battery Backup / UPS	8.18
Block On Call Termination	8.19
Busy Inquiry	8.20
Busy Override	8.39
Busy Queuing - User	8.21
Busy Queuing - User Group	8.22
CO Disconnect Supervision	8.23
CO Pulse/Tone Dialing	8.24
Call Back: External	8.25
Call Back: Internal	8.26
Call Backup - User / User Group	8.27
Call Barring	8.129
Call Broker	8.28
Call Coaching	8.29
Call Conference	8.30
Call Cover	8.31
Call Coverage	8.32
Call Distribution Management	8.33
Call Follow-Me	8.34
Call Forwarding	8.35
Call History Reporting	8.124
Call Hold: Consult	8.36
Call Hold: Exclusive	8.37
Call Hold: System	8.38
Call Intrude	8.39
Call Monitor	8.40
Call Notify	8.41
Call Park	8.42
Call Pickup: Inter-Group	8.43
Call Pickup: Intra-Group	8.44
Call Progress Tones	8.45
Call Priority	8.46
Call Recall	8.47
Call Record	8.48
Call Release	8.49
Call Retrieve	8.50
Call Screen	8.20
Call Transfer: Screened	8.51
Call Transfer: Supervised	8.52
Call Transfer: Unsupervised	8.53
Call Waiting	8.54
Caller ID	8.55
Calling Party Display: Internal	8.56
Caller ID Redial	8.57
Camp-On	8.58
Centrex compatibility	8.59
Class of Service: User	8.60
CTi Interface	8.61
Day/Night Service	8.62
Direct Inward System Access (DISA)	8.63
Direct Station Select (DSS)/BLF	8.64
Direct Line Access	8.95
Direct Trunk Select (DTS)/BLF	8.65
Disconnect Supervision	8.23
Discriminating Ring	8.66
Distinctive Ring	8.67
Do Not Disturb (DND)	8.68
Do Not Disturb (DND) Override	8.69
Duration Timer	8.70
End-to-End Signaling: External	8.71
End-to-End Signaling: Internal	8.72
Exclusive Hold	8.37
Fax Detection	8.73
Flash: Central Office	8.74
Flash - System	8.75
Flexible Call Backup	8.35
Flexible Numbering Plan	8.76
Group Backup	8.27
Group Hunt	8.33
Group Listening	8.77
Group Monitor	8.78
Hands-Free Call	8.79
Hands-Free Block	8.80
Headset Mode	8.81
Home Station	8.82
Hospitality Package	8.83
Intercept Service	8.84
Key Telephone Configuration	8.85
Least Cost Routing	8.86
Line Status Indicators	8.64/8.65
Live Call Screen	8.9
Message Waiting Indication	8.87
Multiple Auto-Attendants	8.88
Multiple Operators	8.89
Multiple Tenant Service	8.90
Music-On-Hold	8.91
Mute	8.92
Name Labeling (System Resources)	8.93
Non-Volatile Configuration Memory	8.94
Off-Hook Preference	8.95
On-Hook Dialing	8.96
Operator Console support	8.97
Operator: Personal	8.98
Operator: System	8.99
PBX Operation	8.100
Page: All Points	8.101
Page: Answer	8.102
Page: Block	8.103
Page: External	8.104
Page: Internal	8.105
Pager Notification	8.106
Password Security	8.107
Port Management: Line and Station	8.108
Privacy Management	8.109
Privacy Release	8.110
Programmable keys: Key Telephone	8.111
Programmable Trunk Timing Parameters	8.112
Redial: Automatic	8.25
Redial: Last Number	8.113
Remote Call Forwarding	8.114
Remote Telephone Notification	8.115
Remote Station Service (RSS)	8.116
Remote System Management (RSM)	8.117
Serial Data Port	8.118
Silent Monitoring	8.119
Software (Windows-based) configuration	8.120
Speakerphone operation	8.121
Speed Dial: System	8.122
Speed Dial: Personal	8.123
Station Message Detail Recording (SMDR)	8.124
Station Relocation	8.125
System Administrator Privileges (SAP)	8.126
System Clock	8.127
System Message Display	8.128
Toll Restriction	8.129
Toll Restriction Override	8.130
Transfer Recall	8.131
Trunk Allocation	8.132
Trunk Routing	8.133
Universal Answer	8.134
Universal Retrieve	8.135
Unsupervised Conference	8.136
User Alias	8.137
Validate Trunk Access	8.12
Voice Mail: Fully Integrated	8.138
Voice mail: Third party	8.139
Volume Control	8.140
Whisper Page	8.141
XOS Upgrade Management	8.142
Zone Paging	8.105

8.2 Access Restriction: Line/User

8.2.1 Description

Access to outside lines and other users on the system are privileges granted by the System Administrator. These access privileges are granted on a user-basis. Users not granted access to lines cannot place outside calls. Users not granted access to other users cannot place internal calls to other users. However, if user Pilot-IDs are defined or if system single digit dialing is defined, they will work regardless of whether or not the User privileges are set.

8.2.2 Interactions

Even when enabled, Access Restriction may be bypassed if any access Pilot-IDs have been defined for the users. For example, to completely block access to trunk lines or other users, Access Restriction must be enabled, and the user Pilot-ID for the trunk must be set to "None". The user then will be unable to seize a trunk.

Access restriction to lines or users may also be set on a temporary basis by the system administrator using **Feature 82/83** (Trunk Access - Deactivate / Activate) and **Feature 84/85** (User to User Access - Deactivate / Activate). Deactivate Access will override any line or extension access rights given to the user. Activate Access will return the access rights back to the user.

8.2.3 Operation

8.2.3.1 For temporary Access Restriction: Line

A. From either a Plexus key telephone or an ordinary phone:

- Activate System Administrator Privileges.
- Enter **F82** + User ID
- To reactivate access, enter **F83** + User ID.

8.2.3.2 For temporary Access Restriction: User to User

A. From either a Plexus key telephone or an ordinary phone:

- Activate System Administrator Privileges.

- Enter **F84** + User ID
- To reactivate access, enter **F85** + User ID.

8.2.4 Configuration

Table 8 - 3 Access Restriction: Line/User

Method	Path	Parameter	Default
Software	User: Privileges	Access Trunks Access User	Granted Granted
Key Telephone	User: User Privileges	Can Access User Can Access Trunk	Granted Granted

8.2.5 Administration

N.A.

8.3 Account Codes: Voluntary

8.3.1 Description

Account Codes offer users a way to tag calls with a numeric label of up to 12 digits in length. These codes are useful for billing and reporting purposes. Account Codes, when utilized, are included with the call record in the SMDR output [see "Station Message Detail Recording (SMDR)" - section 8.124].

8.3.2 Interactions

Plexus systems will only accept the entry of a voluntary account code if the call is first placed in a hold state (System Hold, Exclusive Hold, or Consultation Hold). With a Plexus Key Telephone, the call can be placed on hold automatically by pressing a programmable key defined with the account code feature code (**F88**). This allows a user to press the programmed key during a conversation and immediately enter the account code. Upon entering 1 to 12 digits and the **#** key, the call is automatically retrieved.

Tip
Define a second speed dial programmable key with the user's account code and #. This way the user need only press two keys to enter an account code.

8.3.3 Operation

A. From a Plexus Key Telephone:

- While connected to a trunk, press the preprogrammed **ACCOUNT CODE** feature key + 1 to 12 digit account code + **#**.

or

- Press the **ACCOUNT CODE** feature key + programmable key with the numeric account code and **#** defined. Access trunk and place call.

or

- Press **FLASH** + **F88** + 1 to 12 digit account code + **#**.

or

- Place the call on Exclusive Hold.
- Enter **F88** + 1 to 12 digit account code + **#**.

B. From an Ordinary Phone:

- **FLASH** + **F88** + 1 to 12 digit account code + **#**.

8.3.4 Configuration

N.A.

8.3.5 Administration

N.A.

8.4 Account Codes: Forced

8.4.1 Description

Account Codes: Forced forces the user to tag calls with a numeric label of up to 12 digits in length before making ***any*** external calls. These codes are useful for billing and reporting purposes. Account Codes are included with the call record in the SMDR output [see "Station Message Detail Recording (SMDR)" - section 8.124].

8.4.2 Interactions

With Forced Account Code Entry enabled, a user is not able to access a trunk for placing outgoing calls unless the user first enters an account code. If the user attempts to access a trunk without entering the account code, the system will give an error tone.

The system does not have a database for account codes. When a user enters an account code, any account code from 1 to 12 digits, followed by **#**, will be accepted by the system for trunk access.

8.4.3 Operation

A. From a Plexus Key Telephone:

- Before connecting to a trunk, press the **ACCOUNT CODE** feature key + 1 to 12 digit account code + **#**. Access trunk and place call.

or

- Press the **ACCOUNT CODE** feature key + programmable key with the numeric account code and **#** defined. Access trunk and place call.

or

- Enter **F88** + 1 to 12 digit account code + **#**. Access trunk and place call.

B. From an Ordinary Phone:

- **F88** + up to 12 digits + **#**. Access trunk and place call.

8.4.4 Configuration

Table 8 - 4 Account Codes: Forced

Method	Path	Parameter	Default
Software	User: Features: Options	Forced Account-Code Entry	disabled
Key Telephone	User: User Attributes:	Forced Account Code	No

8.4.5 Administration

Plexus systems will only allow the user access to make an external call after the user enters the Account Code.

8.5 Agent Log Off / Agent Log On

8.5.1 Description

Allows users to log in and out of user groups.

8.5.2 Interactions

Users can log into a user group by dialing **Feature 33**, the user group number, and the user group password. The system will recognize this user as a member of the user group even if this user is not part of the user group in the system configuration. The system will place these users last in line behind currently logged on users for call routing purposes (see "Call Distribution Management" - section 8.33).

To log out of a user group, a user dials **Feature 32** and the user group number. If all users are logged out of a user group, any calls to that user group will follow the user group's Busy Backup/Coverage.

Note
Any user can Log On or Log Off of a user group. The user does not have to be a member of the group in the software configuration to utilize these features.

8.5.3 Operation

8.5.3.1 To Log Off a group

A. From a Plexus Key Telephone or an Ordinary Phone:

- Enter **F32** + user group number + user group password (if applicable).

8.5.3.2 To Log On a group

A. From a Plexus Key Telephone or an Ordinary Phone:

- Enter **F33** + user group number + user group password (if applicable).

8.5.4 Configuration

N.A.

8.5.5 Administration

N.A.

8.6 Agent Not Ready / Agent Ready

8.6.1 Description

Allows users in a group to prevent receiving group calls while staying available for calls that are routed directly to the user's extension

8.6.2 Interactions

User dials **Feature 34** and the user group number to go Agent Not Ready in a group without having to create a busy or DND situation on the user's extension. While Not Ready, the system will read this user as busy for user group call routing purposes but available for calls routed directly to the user's extension.

If all agents are busy or DND and at least one agent is Not Ready, the system will consider the user group to be in the busy state. Calls will Queue or go to the user group's backup/coverage until the user dials **Feature 35** and the user group number to go Ready in the user group.

Note
The Plexus system provides no type of notification of the users Ready / Not Ready. Users should stay aware of their current status.

8.6.3 Operation

8.6.3.1 To go Agent Not Ready

A. From a Plexus Key Telephone or an Ordinary Phone:

- Enter **F34** + user group number.

8.6.3.2 To go Agent Ready

A. From a Plexus Key Telephone or an Ordinary Phone:

- Enter **F35** + user group number.

8.6.4 Configuration

N.A.

8.6.5 Administration

N.A.

8.7 Alarm Clock

8.7.1 Description

The Alarm Clock feature allows users (e.g., front-desk personnel) to program the system to ring other users at a specified time as an alarm, notification, or wake-up call. Access to the Alarm Clock feature is granted the system administrator.

8.7.2 Interactions

Only one alarm may be current for each user. Any attempt to set a second alarm, before the first alarm expires, generates error tone.

Note
A User can only have one Alarm or one Appointment Reminder pending, not both. See “Appointment Reminder” - section 8.10.

8.7.3 Operation

A. From either a Plexus key telephone or an ordinary phone:

- Activate System Administrator Privileges.
- Enter **F46** + User ID + Time (in 24-hour format).
- To deactivate, enter **F45** + User ID.

Note
The time should be entered in a 24-hour clock format. For example, 1:00 a.m. would be 0100, and 2 p.m. would be 1400.

8.7.4 Configuration

If the target user is unavailable (no answer or busy) at the time of an alarm clock call, the system will retry according to the defined retry count and retry interval. The alarm clock call will override DND mode.

The No-Response Cover Pilot receives notification when retry attempts expire with no response. Typically the user issuing the alarm calls is designated as the No-Response Cover Pilot.

Table 8 - 5 Alarm Clock

Method	Path	Parameter	Default
Software	System Parameters: Feature, Alarm Call	No-Response Cover Pilot Retry Count Retry Interval	None 3 2 minutes
Key Telephone	N.A.		

8.7.5 Administration

Refer also to chapter 18, "Hotel / Motel Package".

8.8 Analog Telephone/Device Support

8.8.1 Description

Plexus systems fully support all analog telephones and devices (e.g., modem, FAX machine, credit card scanner, and third-party voice mail adjunct). Analog telephones and devices can be connected to any port on the Analog Extension Interface (AEI). No special adapters are needed.

8.8.2 Interactions

N.A.

8.8.3 Operation

N.A.

8.8.4 Configuration

Table 8 - 6 Analog Telephone/Device Support

Method	Path	Parameter	Default
Software	Extension Port: General/Device/SLT	Port Assignment Location Terminal Type External Voice Adapter Settings DTMF Generation ON-Time DTMF Generation OFF-Time Message Waiting Indication Ring Pattern Int Ring Pattern Ext Ring ON Ring OFF Flash Period Enable Flash Period	None - Telephone - 100 ms 100 ms Audible 60, 20, 20 continuous 1 second 3 seconds enabled 800 ms
Key Telephone	N.A.		

8.8.5 Administration

Should an analog telephone or device be connected to an extension port where a Plexus Key Telephone was previously connected, the port will need to be reset by performing **F99** from the analog telephone. For more information, see chapter 4, "Installation".

8.9 Answering Machine Emulation

8.9.1 Description

The Answering Machine Emulation feature enables a Plexus Key Telephone user to listen to callers as they leave a message in the user's voice mailbox. As the user is screening the call, they can choose to speak to the caller or allow them to continue leaving a message. The user must have a DSS key programmed as **Function 02 -** Answering Machine Emulation.

8.9.2 Interactions

There are two forms of Answering Machine Emulation as follows:

Manual Answering Machine Emulation requires the user to press a programmed key on their key telephone to initiate screening each time a caller is leaving a message. The programmed key flashes when a caller enters the user's mailbox to prompt the user to begin screening.

Automatic Answering Machine Emulation causes the key telephone to automatically go off-hook in speakerphone mode each time a caller is leaving a message. As the caller leaves a message, their voice is broadcast through the speakerphone. The user must pre-press the programmed key. The key will be lit red to indicate that Automatic Answering Machine Emulation is active.

8.9.3 Operation

8.9.3.1 Manual Answering Machine Emulation

Manual Answering Machine Emulation occurs when a caller is in the voice mailbox. The **ANS EMUL** key will flash red when a user is in the voice mailbox.

A. From a Plexus key telephone:

- To listen to the caller leaving a message, press the **ANS EMUL** key.
- To be connected to the caller, press the **ANS EMUL** key again.
- To allow the user to continue leaving a message, press **SPKR**.

8.9.3.2 Automatic Answering Machine Emulation

Automatic screening occurs when there is not a caller in the voice mailbox. The **ANS EMUL** key will light red and each time a caller enters the voice mailbox, the speakerphone will automatically activate enabling active mode Answering Machine Emulation.

A. From a Plexus key telephone:

- Press the **ANS EMUL** key. The key will be lit red to show that Automatic Answering Machine Emulation is active.
- The key telephone will automatically go off-hook in speakerphone mode each time a caller is leaving a message.
- To be connected to the caller, press the **ANS EMUL** key.
- To allow the user to continue leaving a message, press **SPKR**.

8.9.4 Configuration

The Integrated Voice Processor (IVP) must be configured correctly for this feature to work. Refer to the optional IVP section for more information.

8.9.5 Administration

The Plexus Key Telephone must have one of the programmable keys defined with the Answering Machine Emulation function code (**Function 2**) in order to utilize this feature. Answering Machine Emulation cannot be implemented from a single line telephone.

8.10 Appointment Reminder

8.10.1 Description

The Appointment Reminder feature allows a user to program the system to ring them at a specified time as an appointment reminder.

8.10.2 Interactions

Only one appointment may be current for each user. Any attempt to set a second appointment, before the first one expires, generates error tone.

Note
A User can only have one Appointment Reminder or one Alarm pending, not both. See “Alarm Clock” - section 8.7.

If the target user is unavailable (no answer or busy) at the time of an appointment reminder, the system will retry according to the defined retry count and retry interval. Appointment reminders will override DND mode.

8.10.3 Operation

A. From either a Plexus key telephone or an ordinary phone:

Table 8 - 7 Appointment Reminder

To perform this action	Do this
Activate	Enter **F39** + Time (in 24-hour format).
Deactivate	Enter **F38.**

Note
The time should be entered in a 24-hour clock format. For example, 1:00 a.m. would be 0100, and 2 p.m. would be 1400.

8.10.4 Configuration

Note
The Appointment Reminder feature utilizes the same retry count and retry interval as the Alarm Clock feature. The No-Response Cover Pilot does not apply to appointment reminder calls.

If the target user is unavailable (no answer or busy) at the time of an appointment reminder call, the system will retry according to the defined retry count and retry interval. The appointment reminder call will override DND mode.

Table 8 - 8 Appointment Reminder

Method	Path	Parameter	Default
Software	System Parameters: Feature, Alarm Call	Retry Count Retry Interval	3 2 minutes
Key Telephone	N.A.		

8.10.5 Administration

N.A.

8.11 Authorization Management

8.11.1 Description

On Plexus systems, *user* and *user ID* replace the traditional terminology of *extension* and *extension number* and the basic element of authorization is the user. A user is any person or device (e.g., fax, modem, credit card scanner) that places or receives calls on the system. Each user is assigned a unique user ID which is used for system identification. Users log in or sign in to the system with their user ID. When a user logs in or signs in, the system knows where they are located and routes calls accordingly. The advantage of this design is that users are not tied to any particular station.

Note
Plexus systems may be configured such that users have a Home Station to which they are logged into automatically when the system powers up. This option enables the system to be utilized like a more conventional phone system.

8.11.2 Interactions

Users may either log in or sign in to identify themselves to the system and receive their calls at their current location. However, there are differences between log in and sign in. The main difference is that log in applies a user's privileges to the station while sign in does not. For example, a user with no entries in their deny dialing table (see "Toll Restriction" on page 210) can place such calls from any station as long as they are logged in at the station. However, if the user only signs in, he or she is limited to the restrictions applicable to the user that is logged in at the station.

Note
A user can temporarily authorize their external call privileges on another user's extension by using Authorize User (see "Authorize User" - section 8.12).

Sign in is valuable for users that move about an office, working at several different stations on a given day. By signing in at each station, the user ensures that the system will route their calls to them. Sign in is also used for user aliases or personas, enabling a user to sign in different personas to receive different types of calls (see "User Alias" - section 8.137).

8.11.3 Log In and Sign In Basics

- For a user's privileges to apply at a given station, he/she must log in at the station. Stations where no user is logged in are "Not Assigned" and are restricted by the system deny dialing and allow dialing tables (see "Toll Restriction" - section 8.129 and "Authorize User" - section 8.12).
- A user may only log in/sign in at a single station. A subsequent log in/sign in at a second station will automatically log/sign the user out at the initial station.
- Only a single user can be logged in at any given station. A subsequent log in by a different user will automatically log out the initial user.
- Several users may be signed in at the same station. The privileges of the logged in user apply to all of the signed in users.
- A user can be assigned a Home Station where they are automatically logged in each time the system initializes (see "Home Station" - section 8.82 and "Port Management: Line and Station" - section 8.108).
- If a user with an assigned Home Station is not logged in, their calls route to their Home Station.
- Log out does not require a password. Log in only requires a password if one is assigned (see "Password Security" - section 8.107).
- Sign in *and* sign out require a password if one is assigned.

8.11.4 Operation

A. From either a Plexus key telephone or an ordinary phone:

Table 8 - 9 Authorization Management

To perform this action	Do this
Log in	Enter **F31** + User ID + password.
Log out	Enter **F30** + User ID.
Sign in	Enter **F41** + User ID + password.
Sign out	Enter **F40** + User ID + password.

8.11.5 Configuration

N.A.

8.11.6 Administration

User IDs and user privileges need to be assigned to all individuals on the system. Each individual may be assigned more than one user ID. Additional user IDs enable aliases and personas to be utilized for call routing and call distribution purposes (see “User Alias” - section 8.137).

8.12 Authorize User

8.12.1 Description

The Plexus system can allow users to temporarily authorize their external call privileges on another user's extension without logging into that extension (e.g., this can be used for a phone in a common area with no trunk access privileges). This is called Roaming - Voluntary Authorization.

The Plexus system can also be set up to force a user to verify their identity before placing external calls. The user must Validate Trunk Access for placing each external calls during the day and/or night (e.g., this can prevent a phone in a common area from being used to place external calls in the middle of the night). This is called Validate - Forced Authorization.

8.12.2 Interactions

8.12.2.1 Roaming - Voluntary Authorization

This can be used to make external calls on an extension that the current user would be barred or disallowed from making. When the call is terminated, the temporary privileges are removed. The user must dial **Feature 42**, user number, and user Password (if applicable) before accessing an outside line. This must be done before every call.

8.12.2.2 Validate- Forced Authorization

When user-Features-Options-Validate Trunk Access during Day/Night is enabled, the user must dial **Feature 42**, user number, and user Password (if applicable) before accessing an outside line. ***This must be done before placing every external call.***

8.12.3 Operation

A. From a Plexus Key Telephone or an Ordinary Phone:

- Enter **F42** + User # + User Password (if applicable). Then access trunk to place call.

8.12.4 Configuration

8.12.4.1 Roaming - Voluntary Authorization

N.A.

8.12.4.2 Validate- Forced Authorization

Table 8 - 10 Authorize User

Method	Path	Parameter	Default
Software	User: Features, Options	Validate Trunk-Access during Day Validate Trunk-Access at Night	Disabled Disabled
Key Telephone	User: User Attributes	Validate Trunk Day Validate Trunk Night	No No

8.12.5 Administration

N.A.

8.13 Auto-Answer

8.13.1 Description

Auto Answer enables users with Plexus Key Telephones to have their phones answer internal calls automatically. The Key Telephone will immediately go off-hook into speaker phone mode. The user will hear a short stutter tone as the Key Telephone answers the call.

Refer also to “Hands-Free Call” - section 8.79.

8.13.2 Interactions

A transfer from the Integrated Voice Mail (IVP) to an extension port(s) on the Plexus system will not be answered in hands-free mode by a Plexus Key Telephone with auto-answer activated. However, a transfer from a user or an external voice adapter will be answered in hands-free mode by a Plexus Key Telephone with auto-answer activated.

8.13.3 Operation

Auto Answer can be activated / deactivated with a programmed **Auto Answer** key or through the use of **Features 20/21.** To program an **Auto Answer** key, program an available DSS key with **Function 03.**

A. From a Plexus Key Telephone:

- Press the **Auto Answer** key. The key will be red when Auto Answer is activated and off when it is deactivated.

or

- Enter **F21** to activate Auto Answer. Enter **F20** to deactivate Auto Answer.

8.13.4 Configuration

N.A.

8.13.5 Administration

Auto-Answer is activated and deactivated as needed by the user. It is associated with the keyphone, and will remain in effect even if a user logs out.

8.14 Automated-Attendant

8.14.1 Description

The Automated-Attendant feature is available with either of two optional peripheral cards, the Auto-Attendant Interface (AAI) or the Integrated Voice Processor (IVP)-Plexus Integrated Voice Mail. The Auto-Attendant feature provided with the AAI plays menu greetings to callers and allows them to enter digits on their telephone to direct their calls to the desired user/user group or trunks/trunk groups on the Plexus system. The Auto-Attendant feature provided with the IVP is fully programmable allowing for much more advanced call routing capabilities.

Note
Micro-Plexus models are available that incorporate either the AAI or the IVP directly on the motherboard.

8.14.2 Interactions

N.A.

8.14.3 Operation

A. From either a Plexus key telephone or an ordinary phone:

Table 8 - 11 Automated Attendant

To perform this action	Do this
Record a greeting for an AAI card	1. Activate system administrator privileges. 2. Enter **F96** + Auto Attendant ID.
Play a recorded greeting	Dial the Auto Attendant ID. (For information about the IVP Auto Attendant, refer to the IVP manual.)

To record a greeting for an IVP card Auto Attendant, see chapter 9, "Integrated Voice Processor".

8.14.4 Configuration

A logical ID must be defined for the Auto-Attendant resources and the applicable trunks must be routed to the Auto-Attendant ID. Refer to the AAI and IVP sections in the product manual for more information.

8.14.5 Administration

N.A.

8.15 Auto Cover

8.15.1 Description

Auto Cover enables a user to be routed to an unavailable user's coverage without having to follow additional steps. Refer also to "Call Cover" - section 8.31.

8.15.2 Interactions

When a user with Auto Cover enabled calls another user, and that user is unavailable (no answer, busy, DND), the user will be routed to the target user's backup. If the backup is unavailable or not defined, the user will be routed to the target user's coverage.

If a caller without Auto Cover enabled calls another user, and that user is unavailable (no answer, busy, DND), the user will be routed to the target user's backup. If the backup is unavailable, The user will hear unending ringback tone, busy tone, or DND tone, depending on the state of the backup entity (no answer, busy, DND). If no backup is set, the user will hear unending ringback tone, busy tone, or DND tone, depending on the state of the target user (no answer, busy, DND). After receiving these tones, the user will have to press the **MSG** key on a Key Telephone or **Flash + F16** on an ordinary phone to be routed to the target user's coverage (see "Call Cover" - section 8.31).

8.15.3 Operation

N.A.

8.15.4 Configuration

Auto Cover is enabled/disabled for each user.

Table 8 - 12 Auto Cover

Method	Path	Parameter	Default
Software	User: Features, Options	Auto Cover on Internal Calls	Disabled
Key Telephone	User: User Attributes	Auto Cover	No

8.15.5 Administration

N.A.

8.16 Auto Voltage Selection

8.16.1 Description

The Plexus system provides an auto-ranging, auto-switching power supply that automatically selects the appropriate voltage level (110V or 220V) depending on what is available at the installation site.

The power supply provided with the Micro-Plexus system is not an auto-ranging, auto-switching power supply. Therefore, the Micro-Plexus must be purchased from the manufacturer with either a 110V or a 220V power supply.

8.16.2 Interactions

N.A.

8.16.3 Operation

N.A.

8.16.4 Configuration

N.A.

8.16.5 Administration

N.A.

8.17 Automatic Line Selection

8.17.1 Description

Plexus systems can be configured to automatically select an available line for each external outbound call. Users need only enter the system defined trunk-access digit (i.e., a system single digit dialing option with a trunk defined, like “9”) or a particular trunk group ID (e.g., “400”). The system will then identify and connect the user to an available line in that trunk group. Trunk Pilot IDs are assigned on a user-basis. Therefore, each user or user group can be assigned a different group of lines for their outbound calls.

Users may be assigned a single line rather than a trunk group. In this case, the system will always select that particular line when the user enters the trunk-access digit.

Note
Lines may be accessed directly by entering a trunk ID or by pressing an appropriately defined DTS key on a Plexus Key Telephone.

8.17.2 Interactions

N.A.

8.17.3 Operation

N.A.

8.17.4 Configuration

Table 8 - 13 Automatic Line Selection

Method	Path	Parameter	Default
Software	System Parameters: Dial Plan User: Assignment: Group Access	Access Digit for Trunk Pilot-ID for Trunk	9 None
Key Telephone	User: User Attributes	Trunk Pilot ID	0

8.17.5 Administration

It is recommended that a portion of the available programmable keys on Plexus Key Telephones be used for monitoring trunks. Programmable keys defined as such are referred to as DTS (Direct Trunk Select) keys and may be used to quickly access available trunks.

Users that will be making external calls must be granted trunk access privileges to select a particular trunk line. Otherwise, only trunk Pilot IDs and system single digit trunk access will be allowed. See “Access Restriction: Line/User” - section 8.2.

8.18 Battery Backup / UPS

8.18.1 Description

Plexus Systems provide an internal Uninteruptable Power Supply (UPS). Batteries are not included and must be added to the Plexus for the UPS to work properly.

It is strongly recommended that the Plexus systems internal UPS be connected to a 24-volt battery source in order to maintain system operation during a power failure.

8.18.2 Interactions

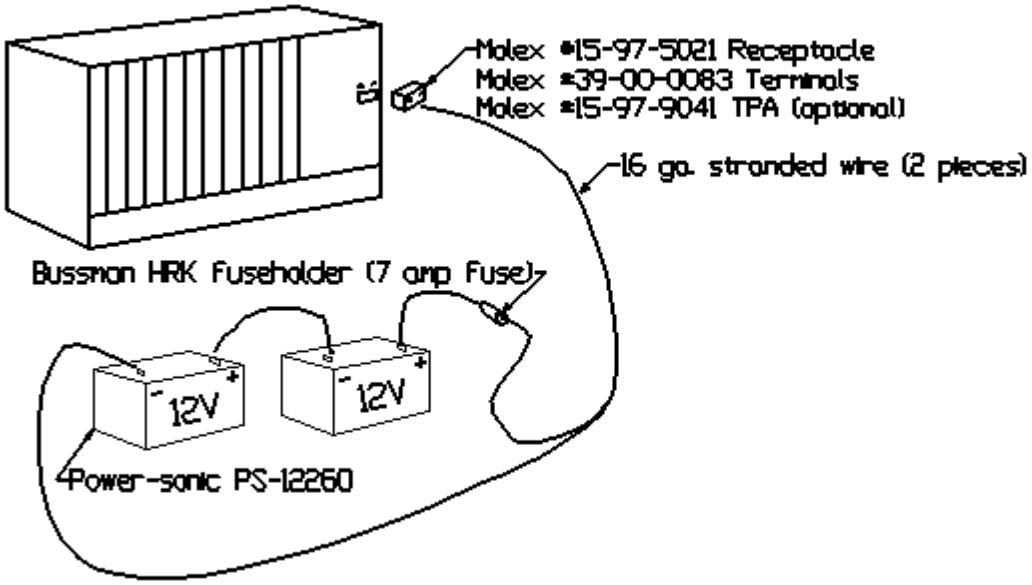
The battery backup / UPS will provide instant power to the Plexus system in cases on power outages or power drops. The system will continue to operate normally. No indication will be given that the system is operating under backup power. The time length of power provided will vary based on the types of batteries installed (see chapter 4, "Installation")

8.18.3 Operation

N.A.

8.18.4 Configuration

The recommended configuration consists of two 12-volt batteries connected in series as shown below.



The length of time that the system can operate on battery power is primarily determined by the characteristics of the particular battery being used. The battery draw of the Plexus systems is 6 A @ 24V max or 2.5 A @ 24V nominal. With a 24V, 20-26 Ah (Plexus) or 10 Ah (Micro-Plexus) battery, the system can operate for 2-4 hours at maximum draw or 6-8 hours at nominal draw.

8.19 Block On Call Termination

8.19.1 Description

When a user is on an internal or external and the other party disconnects, block on call termination provides a busy signal to the user rather than the normal dial tone. This feature emulates the disconnect tone that several phone services send after the remote party disconnects from a call.

8.19.2 Interactions

When a user is disconnected to a call and remains off hook, the system will normally send a new dial tone. If block on call termination is enabled, the user will only receive an error tone.

8.19.3 Operation

N.A.

8.19.4 Configuration

Table 8 - 14 Block on Call Termination

Method	Path	Parameter	Default
Software	User: Features: Options	Block on Call Termination	Disabled
Key Telephone	User: User Attributes	Block on Call Terminate	No

8.19.5 Administration

N.A.

8.20 Busy Inquiry

Busy Inquiry, also known as Call Screen, is typically used to determine who is holding on a particular line, who is using a particular line, or who is a user connected to.

Plexus Key Telephone users (display model only) can view call information associated with any busy user, in-use trunk, or user group for which they have a defined DSS or DTS key. Available information may include the user ID, trunk ID, Caller ID information, or user group information. The information will be shown on the display window of the Plexus Key Telephone.

8.20.1 Interactions

The information available for busy users and in use trunks will vary based on a users privileges.

If a user without administrate privilege does a Busy Inquiry, the user will be able to see user ID(s) and trunk ID for busy users and trunks, and associated user ID and Caller ID information on trunks that have been placed on System Hold.

If a user with administrate privilege does a Busy Inquiry, the user will be able to see user ID(s) and Caller ID for busy users and trunks, and associated user ID and Caller ID information on trunks that have been placed on System Hold.

Any user can access information about user groups. Pressing **SCREEN** and the **DSS** key mapped to a user group displays: _Queued (number of people currently on hold in the Queue), _Min (the time in minutes the oldest call has been in the Queue), _Mbrs (the number of users logged on to the group), _+ (total calls answered), _- (Queued calls that disconnected), and _? (calls that pressed '#' to leave a message). If the user group does not have Hold Queue enabled, all Hold Queue information will remain at 0.

Busy Inquiry is not available for DTS, DSS, or Call Broker keys associated with call conference parties.

Note
If a user does a Busy Inquiry on a user that is off hook but not connected to another party, no information will be displayed.

8.20.2 Operation

A. The Busy Inquiry Feature is available from a Plexus Key Telephone with display only.

- Press the **Screen** key.
- Press the mapped **DSS** key associated with a user group or busy user.

or

- Press the **Screen** key.
- Press the mapped **DTS** key associated with an in use trunk.

8.20.3 Configuration

N.A.

8.20.4 Administration

It is recommended that a portion of the available programmable keys on Plexus Key Telephones be used for monitoring trunks, users, and user groups. Programmable keys defined as such are referred to as **DTS** (Direct Trunk Select) or **DSS** (Direct Station Select) keys.

8.21 Busy Queuing - User

8.21.1 Description

The Busy Queuing - User feature places external calls to a busy user into a temporary Hold Queue. The held calls are routed to user when the user becomes available. The system can give a reassurance message to the held caller or can do a temporary routing of the call to a system entity (user or user group) to provide a reassurance to the caller.

8.21.2 Interactions

The user determines whether or not calls can be placed in user's Hold Queue (see Operation).

Calls can be placed in the Hold Queue for a busy user for the duration of the user's Exclusive Hold timeout. When a call enters the user's queue, the user will hear a tone alerting of the held call. Once the Exclusive Hold timeout expires, the call is routed to the user's backup or coverage entities (see"Call Backup - User / User Group" - section 8.27 and "Call Coverage" - section 8.32).

Busy Queuing can be set up such that the queue is intercepted periodically with a default system message, a custom message, or a connection to another user or user group on the system. See "Intercept Service" on page 145 for more information.

Calls are placed in the Hold Queue in a first-in first-out basis. This means that even if a reassurance message is being played for a caller, if that caller was the first caller placed in the Hold Queue, that caller will instantly be routed to the user when the user becomes available.

8.21.3 Operation

A. From a Plexus Key Telephone:

- For a one touch Queuing enabled/disabled button, a **DSS** Key must be programmed as **Function 05**. A solid red light indicates that external call Busy Queuing is enabled. No light indicates that external call Busy Queuing is disabled
- A user can also dial **F61** to enable their external call Busy Queuing and **F60** to disable their external call Busy Queuing.

B. From an Ordinary Phone:

- A user dials **F61** to enable their external call Busy Queuing and **F60** to disable their external call Busy Queuing.

8.21.4 Configuration

Table 8 - 15 Busy Queuing User

Method	Path	Parameter	Default
Software	User: Features: Assignments	Timeout on Exclusive Hold	5 minutes
Key Telephone	User: User Attributes	X-hold timeout	5 minutes

8.21.5 Administration

N.A.

8.22 Busy Queuing - User Group

8.22.1 Description

The Busy Queuing - User Group feature places external calls to a busy user group into a temporary Hold Queue. The held calls are routed to user when any user in the user group becomes available. The system can give a reassurance message to the held caller or can do a temporary routing of the call to a system entity (user or user group) to provide a reassurance to the caller.

8.22.2 Interactions

Calls can be queued for a busy user group (e.g., Sales, Technical Support) for the duration of the group's hold queue timeout. Once the hold queue timeout expires, the call is routed to the user group's backup or coverage entities (see"Call Backup - User / User Group" - section 8.27 and "Call Coverage" - section 8.32).

Busy Queuing can be set up such that the queue is intercepted periodically with a default system message, a custom message, or a connection to another user or user group on the system. See "Intercept Service" - section 8.84 for more information.

Calls are placed in the Hold Queue in a first-in first-out basis. This means that even if a reassurance message is being played for a caller, if that caller was the first caller placed in the Hold Queue, that caller will instantly be routed to the user group when a user becomes available.

The system can notify a user if there is a call holding in the user group's Hold Queue. The user must have a **DSS** key mapped as monitoring the user group and must have Call Notify enabled. When a call is in the Hold Queue, the user will receive a stutter tone through the handset or speaker of a Plexus Key Telephone (see "Call Notify" - section 8.41).

8.22.3 Operation

N.A.

8.22.4 Configuration

To enable busy queuing for a user group, enable the hold queue timeout and establish coverage (see “Call Coverage” - section 8.32) and, if desired, backup (see “Call Backup - User / User Group” - section 8.27).

Table 8 - 16 Busy Queuing - User Group

Method	Path	Parameter	Default
Software	User Group: Distribution	Hold Queue Timeout (Enabled)	Disabled/ 2 minutes
Key Telephone	User Group: User Group Attributes	Queuing Enabled Hold Queue to (m)	No 2 minutes

8.22.5 Administration

N.A.

8.23 CO Disconnect Supervision

8.23.1 Description

Plexus systems are capable of detecting several types of Central Office disconnect signals including Clear Forward, Polarity Reversal, and Disconnect Tones.

Refer also to “Programmable Trunk Timing Parameters” - section 8.112.

8.23.2 Interactions

N.A.

8.23.3 Operation

N.A.

8.23.4 Configuration

The Disconnect Type must first be selected. If Clear Forward or Polarity Reversal are selected, the associated timing parameter may also need to be modified. If Disconnect Tone is selected, the necessary filter must be installed and activated (see “Call Progress Tones” - section 8.45).

Table 8 - 17 CO Disconnect Supervision

Method	Path	Parameter	Default
Software	Trunk Port: Features System Parameters: Call Progress, Detector	Disconnect Type Seize Settling (SS) (Central) Office Settings (OS) CF (Clear Forwarding) PR (Polarity Reversal) Tone Type, Disconnect	Clear Forward 1000 ms 500 ms 600 ms 0 ms -
Key Telephone	N.A.		

Note
Call Progress filters can only be configured through software configuration. Should the filters provided not meet the needs at a particular installation, email a request for additional filters to bbstechs@bbstelecom.com. (**Specify the frequency range you need.**)

8.23.5 Administration

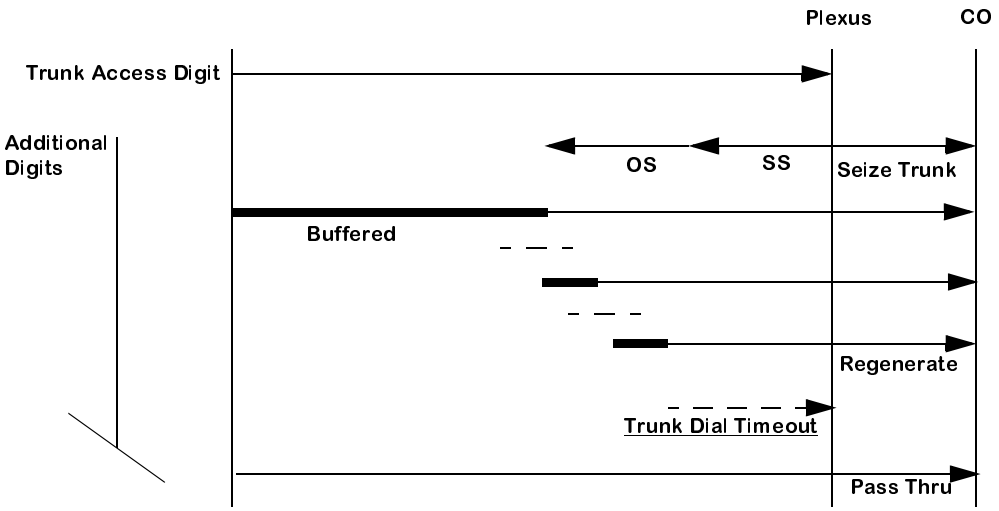
There is a set of two pins for each trunk port on an Analog Trunk Interface (ATI) card. Each set of pins is labeled with a "JP" and a number. At sites where clear forwarding is being utilized, the pins should be jumpered for a closed position. At sites where polarity reversal is being utilized, the pins should *not* be jumpered for an open position.

Note
All trunk ports on a single system should utilize the same type of disconnect supervision.

8.24 CO Pulse/Tone Dialing

8.24.1 Description

Each trunk port on a Plexus system can be configured for either pulse or tone dialing. The type of dialing at the extension is independent of the dialing generated at the trunk port. Therefore, a pulse dialing ordinary telephone may dial through a tone dialing trunk port. This is accomplished through the buffering and regeneration of dialed digits as shown below.



When a user dials the trunk-access digit, the system seizes a trunk and begins to buffer any additional digits. As soon as the Seize Settling (SS) and the Central Office Settling (OS) periods expire, the buffer is emptied and the dialed digits are regenerated either as pulses or tones, depending on how the trunk port is configured. As soon as the buffer is emptied, the Trunk Dial Timeout period commences. If additional digits are dialed before the Trunk Dial Timeout expires, they are buffered and regenerated. Each time the buffer is emptied, the Trunk Dial Timeout period commences again. As soon as the Trunk Dial Timeout expires without additional digits being dialed, the buffer/regeneration process ceases. Any digits dialed at this point are passed through to the Central Office as dialed at the extension.

Refer also to “Programmable Trunk Timing Parameters” - section 8.112.

8.24.2 Interactions

When a call is being routed using Least Cost Routing or when a call is being routed on a digital trunk, the system will wait until the user has completed dialing before routing the call. After the user has physically completed dialing, the system waits until the Network Dial TO has expired to ensure that the user has finished before routing the call. To reduce the delay between the completion of dialing and the actual routing of the call, users can dial '**#**' to terminate dialing and to let the system know that dialing is complete. The system will instantly route the call without waiting for the Network Dial TO to expire.

8.24.3 Operation

N.A.

8.24.4 Configuration

Table 8 - 18 CO Pulse/Tone Dialing

Method	Path	Parameter	Default
Software	Trunk Port: Dialing	Dial Mode Dial-Timeout Pulse Timing Inter-Digit Pulse Timing Make Pulse Timing Break DTMF Timing On DTMF Timing Off	Tone 5000 ms 500 ms 33 ms 66 ms 100 ms 100 ms
Key Telephone	N.A.		

8.24.5 Administration

Extension ports on the Analog Extension Interface (AEI) support both pulse and tone dialing. The system automatically identifies the type of dialing being used and configures the port accordingly.

8.25 Call Back: External

8.25.1 Description

Call Back: External allows a user to instruct the system to automatically redial a busy external number according to a defined retry count and interval. When the system reaches the number in an available state (i.e., detects ringback tone), it recalls the responsible user and connects them to the call.

Note
The feature only applies to external calls.

8.25.2 Interactions

If the responsible user is unavailable at the time of a Call Back: External recall, the Call Back: External terminates.

The system utilizes the same line that the original call was placed on for all retry attempts. Once the Call Back: External feature code is issued, the line is effectively reserved to perform the defined number of retries.

8.25.3 Operation

A. From a Plexus key telephone:

- After receiving a busy signal, press the **CALL BACK** key.

B. From an ordinary phone:

- After receiving a busy signal, **FLASH** + **F18**.

8.25.4 Configuration

The Retry Count is the number of times the system will attempt to reach the busy number. The Retry Interval is the period of time between retry attempts.

Call Back: External relies on installing a filter profile for Busy tone and assigning this filter to a tone detector. The filters and detectors are not installed or activated by default. Once identified by the detector, the system will redial a busy external number according to a defined retry count and interval. When the system reaches the number in an available state (i.e., detects ringback tone), it recalls the responsible user and connects them to the call.

Table 8 - 19 Call Back: External

Method	Path	Parameter	Default
Software	System Parameters: Feature, Limits	Auto-Redial Retry Count Auto-Redial Retry Interval	10 60 seconds
Key Telephone	N.A.		

8.25.5 Administration

The Busy Filter Detect on the DXP card must be set through the Windows Administrator. See chapter 6, "Software Configuration" for more information.

8.26 Call Back: Internal

8.26.1 Description

The Call Back feature allows users to request a call back from the system when an unavailable entity becomes available. Internal call back may be invoked when trying to reach a busy user, user group, trunk, or trunk group. As soon as the unavailable entity becomes available, the system rings the user and then completes their connection. If the user is unavailable (busy) when the call back occurs, the call back is canceled. If the user is in DND (Do Not Disturb), the call back will override the DND.

Notes
If all of the trunks in a user's assigned trunk group are being used, the user will hear busy tone upon entering the trunk-access digit. If the user requests a call back, the system will ring them when one of the trunks in the trunk group becomes available.

8.26.2 Interactions

N.A.

8.26.3 Operation

A. From a Plexus Key Telephone:

- After receiving a busy signal, press the **CALL-BACK** key**.**

or

- After receiving a busy signal, press the **FLASH** key and **F18**.

B. From an Ordinary Phone:

- After receiving a busy signal, **FLASH** + **F18.**

8.26.4 Configuration

N.A.

8.26.5 Administration

N.A.

8.27 Call Backup - User / User Group

8.27.1 Description

Call Backup provides an intermediate level of coverage for users and user groups. In cases where the user is unavailable or there is no available agent in the target user group, calls route to the applicable backup entity (if defined) before routing to coverage. Different backup entities (i.e., user, user group, voice ID, external operator, answering service) can be defined for each unavailable state.

8.27.2 Interactions

If a voice ID is entered as a backup entity, the call is routed to the route point associated with the voice ID and enters the corresponding Auto Attendant structure as if routed there directly. Refer to the optional IVP section for more information.

Call Backup may be used without or without busy queuing enabled. If busy queuing is enabled, calls are queued when the user or all agents in a group are busy. If the hold queue timeout expires, calls route to the applicable backup entity.

8.27.3 Operation

N.A.

8.27.4 Configuration

Table 8 - 20 Call Backup - User / User Group

Method	Path	Parameter	Default
Software	User: Backup:	Busy	None
		DND	None
		No Answer	None
	User Group: Backup	Busy	None
		DND	None
		No Answer	None
Key Telephone	User: User Attributes	No Ans Backup	0
		Busy Backup	0
	User Group: User Grp Attrib	DND Backup	0
		No Ans Backup	0
		Busy Backup	0
		DND Backup	0

8.27.5 Administration

N.A.

8.28 Call Broker

Plexus Key Telephones (PVT-30/30D and PDT-30D) have three call appearance keys which automatically map to calls for which there is no associated DTS or DSS key. For example, if a call is received on trunk 210 and trunks 201-209 are the only ones with assigned DTS keys, the call appears on one of the call appearance keys. These keys, known as Call Broker keys, allow a Plexus Key Telephone user to manage their calls and switch between calls using the same methods available when there is an associated DTS or DSS key (e.g., pressing the key to place the call on Exclusive Hold).

Notes
The Call Broker keys are labeled **CALL 1**, **CALL 2**, and **CALL 3**.

Note
* On version 1.00 and version 1.01 version keyphones, If a call on a **Call Broker** key is placed on hold using the **Hold** key, the call will no longer be mapped on the user's Keyphone. It is recommended that these users use the **Call Broker** key for placing calls on hold.

8.28.1 Interactions

N.A.

8.28.2 Operation

A. From a Plexus Key Telephone:

For Internal Calls:

1. When calling an extension that does not have an assigned DTS or DSS key, the call will associate with one of the Call Broker keys (Call 1, Call 2, or Call 3).
2. To place the call on hold, press its **CALL BROKER** key or the **Hold** key. The key will flash to indicate that it has a call on hold*.
3. To retrieve the call from hold, press its flashing **CALL BROKER** key.

4 When retrieving another **CALL BROKER** key, place the first call on hold before selecting the second call's **CALL BROKER** key. Otherwise, the first call will be disconnected.

For External Calls:

1 When calling an external number from an extension that does not have an assigned DTS or DSS key, the call will associate with one of the Call Broker keys (Call 1, Call 2, or Call 3).

2 To place the call on hold, press its **CALL BROKER** key or the **Hold** key. The key will flash to indicate that it has a call on hold*.

3 To retrieve the call from hold, press its flashing **CALL BROKER** key.

4 To retrieve another call from hold, press its **CALL BROKER** key. The first call will automatically be placed on hold.

Note
* On 1.00 and 1.01 version keyphones, If a call on a **Call Broker** key is placed on hold using the **Hold** key, the call will no longer be mapped on the user's Keyphone. It is recommended that these users use the **Call Broker** key for placing calls on hold.

8.28.3 Configuration

N.A.

8.28.4 Administration

It is recommended that a portion of the available programmable keys on Plexus Key Telephones be used for monitoring trunks and users. Programmable keys defined as such are referred to as **DTS** (Direct Trunk Select) or **DSS** (Direct Station Select) keys.

8.29 Call Coaching

8.29.1 Description

Call Coaching allows a user to intrude on a target user's call to coach the target user through the call. The user will be able to hear the target user's conversation, but the remote party will be unaware of the intrusion and will not hear the calling user's coaching.

8.29.2 Interactions

The calling user will be able to hear the target user's conversation, but the remote party will be unaware of the intrusion and will not hear the calling user's coaching.

Note
The target user ***must*** be on a ***Digital*** Plexus Key Telephone (PDT-30D or PDT-22D) for Call Coaching to be able to work properly.If a user attempts this Feature on an Analog Keyphone, the results will not be to standard or guaranteed.

8.29.3 Operation

A. From a Digital Plexus Key Telephone:

- On receiving a busy signal, press the **DSS** key programmed as **F15.**

or

- On receiving a busy signal, press **FLASH** + **F15.**

8.29.4 Configuration

To use Call Coaching, a user must have Intrude privileges for the target user.

Table 8 - 21 Call Coaching

Method	Path	Parameter	Default
Software	User: Privileges or User: Override: Intrude	Intrude All Name	Not Granted None
Key Telephone	User: User Privileges	Can Intrude All	No

8.29.5 Administration

N.A.

8.30 Call Conference

8.30.1 Description

A Call Conference is a call that consists of more than two parties. Plexus systems support call conferences of up to six parties made up of users and/or external parties. It is even possible, with special privileges, for a user to set up a call conference with several external parties and then drop out of the conference allowing the external parties to continue (see “Unsupervised Conference” - section 8.136).

8.30.2 Interactions

Even though call appearances are available on a Plexus Key Telephone for each party in a call conference, the system still recognizes the call conference as a single call. Therefore, all parties are placed on hold simultaneously. For example, if a user presses either a Call Broker key or a DTS key associated with a conference party, all of the parties to the conference will be placed on Exclusive hold. Likewise, retrieving one party, retrieves the entire conference.

The Busy Inquiry feature is not available for DTS, DSS, or Call Broker keys associated with call conference parties.

Enabling Call Waiting on an ordinary phone interferes with the ability to call conference.

The displays on any Plexus Key Telephones will indicate “Conferenced”.

Individual parties may be dropped (released) from a call conference by using the Release Key. Refer to “Call Release” on page 92 for more information.

8.30.3 Operation

A. From a Plexus Key Telephone:

- While conversing with another party (or conference), press the **CONF** key.
- Dial a User ID or Phone #.
- Press **CONF**.

You will now be conferenced with the original party(ies) and the newly called party.

If an attempted party is busy, not answering, or in Do Not Disturb (DND) mode, press **RELEASE** to return to system dial tone.

You can then try another party or retrieve the conference from **Call Hold: Consult** by pressing on any of the green blinking keys.

B. From an Ordinary Phone:

- While conversing with another party (or conference), press the **FLASH** key.
- Dial a User ID or Phone #.
- **FLASH**.

You will now be conferenced with the original party(ies) and the newly called party.

If an attempted party is busy, not answering, or in Do Not Disturb (DND) mode, **FLASH** to return to system dial tone.

You can then try another party or retrieve the conference from **Call Hold: Consult** with a **FLASH**.

8.30.4 Configuration

N.A.

8.30.5 Administration

N.A.

8.31 Call Cover

8.31.1 Description

When a user receives no answer, a busy signal, or a Do Not Disturb (DND) indication on an internal call, the Call Cover feature may be utilized to reach the unavailable user's call coverage (e.g., voice mailbox, operator).

Notes
The Call Cover feature is only available for internal calls and is not available from an ordinary phone on a no answer response.

Refer also to "Call Coverage" - section 8.32.

8.31.2 Interactions

N.A.

8.31.3 Operation

A. From a Plexus Key Telephone:

- After receiving a no answer/busy/DND response from a user, press **MSG.**

or

- After receiving a no answer/busy/DND response from a user, press **FLASH** + **F16**.

B. From an Ordinary Phone:

- After receiving a busy or DND response from a user, **FLASH** + **F16**.

Note
Call Cover is not available for no answer responses on ordinary phones.

8.31.4 Configuration

N.A.

8.31.5 Administration

N.A.

8.32 Call Coverage

8.32.1 Description

Call Coverage refers to a system feature that routes calls to an alternative entity, Internal (e.g., receptionist, voice mailbox, colleague) or External (e.g., answering service, remote answering machine), when the target user or user group is unavailable. There are three unavailable states recognized by the system. These are; No Answer, Busy, and DND. Coverage parameters exist for all three unavailable states. Therefore, different coverage entities may be defined for each different unavailable state.

8.32.2 Interactions

8.32.2.1 User

- No Answer refers to cases where calls placed or transferred to the user go unanswered.
- Busy refers to cases where the user is already on a call and therefore cannot receive additional calls and Busy Queuing is not enabled. When Busy Queuing is enabled and the Exclusive Hold Timeout has expired, the system considers the user busy (see “Busy Queuing - User” - section 8.21). (Also see “Call Waiting” - section 8.54).
- DND refers to cases where the user has placed their station in Do Not Disturb mode preventing calls from reaching their station.

8.32.2.2 User Group

- No Answer refers to cases where calls placed or transferred to the user group go unanswered. Note that the system need only identify a single agent to ring in order for the call to go unanswered. Therefore, it is possible for a call to go to No Answer coverage even in cases where there are several busy agents and/or agents in Do Not Disturb mode.
- Busy refers to cases where all agents are either busy or in Do Not Disturb mode and Busy Queuing is not enabled. When Busy Queuing is enabled and the Queue Timeout has expired, the system considers the user group busy. Note that the system need only identify a single busy agent for the call to go to Busy coverage. Therefore, it is possible for a call to go to Busy coverage even in cases where there are several agents in Do Not Disturb mode.

- DND refers to cases where all agents in a user group are in Do Not Disturb mode.

Note
Calls routed to an unavailable coverage entity (pilot) are dropped. Therefore, it is very important to carefully select pilots.

8.32.3 Operation

N.A.

8.32.4 Configuration

Table 8 - 22 Call Coverage

Method	Path	Parameter	Default
Software	User: Coverage: User Group: Coverage	Internal / External* Busy DND No Answer External Dial #	Internal None None None blank
Key Telephone	User: Attributes User Group: Attributes	No Answer Cover* No Answer Dial # Busy Cover* Busy Cover Dial # DND Cover* DND Cover Dial #	0 Blank 0 Blank 0 Blank

Note
If Coverage is set as external, an external phone number must be entered. If calls are routed to an external phone number that is not available (e.g., no answer or busy) or if an external phone number has not been entered, the system will terminate the call.

8.32.5 Administration

N.A.

8.33 Call Distribution Management

8.33.1 Description

Call distribution management defines the manner in which calls are distributed among members of a user group. Plexus systems support six different distribution methods. They are Linear, Circular, Least Busy, Multiple, Manual, and Single Shot.

8.33.1.1 Linear

Attempts to distribute all calls to a single agent. If the agent is unavailable (No Answer, Busy, DND) the call is distributed to the next available agent according to the order in which they are listed in the user group.

8.33.1.2 Circular

Distributes calls to all available agents in the user group in a circular fashion according to the order in which the users are listed.

8.33.1.3 Least Busy

The system calculates a busy value for each user in the user group and routes each subsequent incoming call to the least busy user. When new users log in, their busy value is set equal to that of the most busy agent.

8.33.1.4 Multiple

Simultaneously rings every available agent in the user group for the duration of the No Answer Timeout. Should an agent hang up during a simultaneous ringing, their phone will begin ringing.

8.33.1.5 Manual

Immediately places the call in the Hold Queue of the group, allowing available users to grab the call as the users are able. When a call is in the Queue, users with that group monitored on their keyphone will be able to retrieve the call by pressing the flashing DSS Key that is mapped to the user group.

8.33.1.6 Single Shot

For future release.

8.33.2 Interactions

N.A.

8.33.3 Operation

N.A.

8.33.4 Configuration

The No-Answer Timeout is the period that a call will ring an agent before rolling to the next agent or routing to backup, if defined, (see “Call Backup - User / User Group” - section 8.27) or coverage (see “Call Coverage” - section 8.32). Wrap-Up Time refers to the delay between when an agent becomes free and the time when the next call in the queue enters the hunt for the free agent.

Table 8 - 23 Call Distribution Management

Method	Path	Parameter	Default
Software	User Group: Distribution	Distribution Method No-Answer Timeout Wrap-Up Time	Linear 20 seconds 1 minute
Key Telephone	User Group: User Group Attributes	Hunt Method No Answer Timeout Wrap-Up Time	Linear 20 seconds 1 minute

8.33.5 Administration

A user must be either logged in or signed in as a member of a group to receive calls.

8.34 Call Follow-Me

8.34.1 Description

Call Follow-Me is a form of internal Call Forwarding which can be activated by a user from a station other than the station at which they are logged in. When activated, all calls for the user are immediately forwarded to their current location.

Refer also to"Call Forwarding" - section 8.35 and "User Alias" - section 8.137.

Note
The Version 1.00 keyphone may not indicate that calls are being diverted or forwarded when the user logs back onto to them.

8.34.2 Interactions

If the station to which the user's calls are forwarded is unavailable (i.e., No Answer, Busy, DND), calls will still route to the user's applicable backup or coverage entity.

8.34.3 Operation

A. From either a Plexus key telephone or an ordinary phone:

Table 8 - 24 Call Follow Me

To perform this action	Do this
Activate from any station	Enter **F67** + User ID.
Deactivate from any station	Enter **F66** + User ID

8.34.4 Configuration

Table 8 - 25 Call Follow Me

Method	Path	Parameter	Default
Software	User: Privileges	Internal FWD	Granted
Key Telephone	User: Privileges	Can Internal FWD	Yes

8.34.5 Administration

N.A.

8.35 Call Forwarding

8.35.1 Description

Call Forwarding enables users to temporarily define their backup. The backup can be set to immediately or selectively forward calls to an internal entity or an off-site telephone.

8.35.1.1 Call Forwarding Immediate (All)

All calls are forwarded immediately.

8.35.1.2 Call Forwarding on No Answer

Calls are forwarded only if they go unanswered (according to the user's Timeout on No Answer).

8.35.1.3 Call Forwarding on Busy

Calls are forwarded only when the user is busy.

8.35.1.4 Call Forwarding on DND

Calls are forwarded only when the user is in Do Not Disturb mode.

8.35.2 Interactions

Internal Call Forwarding is subject to one restriction. The user must have the privilege Internal FWD enabled.

External Call Forwarding is subject to three restrictions. The first is a user privilege, External FWD must be enabled. The second is the user's deny dialing and allow dialing tables (see "Toll Restriction" - section 8.129). If the number the call is being forwarded to is on the deny dial list, the call forwarding will fail. The third is the Unattended Call Duration. All trunk-to-trunk calls made on the system terminate according to this timeout. The system utilizes a line according to the user's assigned trunk Pilot-ID (see "Trunk Allocation" - section 8.132) to forward the call. If no trunk assignment is made, the call forward fails. The user must specify a trunk access digit or a trunk ID in the dial string for external Call Forwarding.

If voicemail attempts a supervised transfer to a user that is externally forwarded using **F62**, the call will not be completed. Instead, it will be routed according to the busy option settings for mailbox. See chapter 9, "Integrated Voice Processor" to setup external forwarding through the voicemail.

If a user (e.g., receptionist or operator) transfers an external caller to an extension that is being forwarded, the target user must have the Unattend Trunk privilege enabled for the call to be routed successfully.

If the Call Forwarding fails, the call will be routed to the user's or user group's coverage entity.

Note
If a user is assigned to a port and logs out of a station, Call Forwarding will stay in effect until the user logs in and disables Call Forwarding. If a user is not assigned to a port and enables call forwarding, when the user logs out of a station the Call Forwarding will terminate.

8.35.3 Operation

From either a Plexus key telephone or an ordinary phone:

8.35.3.1 Call Forwarding Immediate (All)

- To activate Call Forwarding Immediate, enter **F62** + the target number + **#**.
- To deactivate Call Forwarding Immediate, enter **F62** + **#**.

8.35.3.2 Call Forwarding on No Answer

- To activate Call Forwarding on No Answer, enter **F63** + the target number + **#**.
- To deactivate Call Forwarding on No Answer, enter **F63** + **#**.

8.35.3.3 Call Forwarding on Busy

- To activate Call Forwarding on Busy, enter **F64** + the target number + **#**.
- To deactivate Call Forwarding on Busy, enter **F64** + **#**.

8.35.3.4 Call Forwarding on DND

- To activate Call Forwarding on DND, enter **F65** + the target number + **#**.
- To deactivate Call Forwarding on DND, enter **F65** + **#**.

Note - **External Call Forwarding**
To forward the call to an external number, a trunk must be accessed. Enter the user's trunk access digit (default = 9) or a trunk or trunk group ID + the target number (E.g., 9 (or trunk ID) + 555-1212 + #)

8.35.4 Configuration

Table 8 - 26 Call Forwarding

Method	Path	Parameter	Default
Software	User: Privileges User: Deny Dialing User: Allow Dialing System Parameters: Feature, Timeouts User: Assignment, Group-Access	External FWD Number Number Unattended Call Duration Pilot ID for Trunk	Not Granted - - 15 minutes None
Key Telephone	User: Privileges User: Dial Deny User: Dial Allow System Parameters	Can External FWD Number Number Unattended Call Duration	No -- -- 15 minutes

8.35.5 Administration

N.A.

8.36 Call Hold: Consult

8.36.1 Description

Consult Hold is a temporary, exclusive hold automatically invoked by the system when calls are in the process of being transferred or conferenced.

8.36.2 Interactions

The process of transferring calls, adding new calls to a call conference, or implementing Feature Codes by using **FLASH** automatically places any active calls on Consult Hold. Calls are automatically retrieved from Consult Hold upon completion of the transfer, successful addition of a new call to the call conference, or implementation of the Feature Code. Calls may also be manually retrieved from Consult Hold. If a call is not retrieved from Consult Hold (either automatically or manually), it will immediately recall the responsible user if the user hangs up before the Timeout on Exclusive Hold expires. Should a call remain on Consult Hold for a period that exceeds the Timeout on Exclusive Hold, the call attempts to recall the responsible user. If the responsible user is unavailable, the call routes to the responsible user's applicable Backup entity. If the Backup entity is unavailable or not defined, the call routes to the applicable coverage entity for the responsible user.

Note
Music-On-Hold is not broadcast to calls on Consult Hold.

8.36.3 Operation

A. From a Plexus key telephone:

Calls will be automatically placed in this state when utilizing **Call Transfer, Call Conference**, or implementing Feature Codes by using **FLASH**.

B. From an ordinary phone:

- To place a call on Call Hold: Consult **FLASH**.
- To retrieve a call on Call Hold: Consult **FLASH**.

8.36.4 Configuration

Table 8 - 27 System Hold: Consult

Method	Path	Parameter	Default
Software	User: Features, Timeouts	Timeout on Exclusive Hold	5 minutes
Key Telephone	User: User Attributes	X-hold Timeout	5 minutes

8.36.5 Administration

N.A.

8.37 Call Hold: Exclusive

8.37.1 Description

Exclusive Hold enables a user to place calls on hold in such a way that no other user on the system can retrieve the call by pressing the mapped **DSS/DTS** key. Other users on the system will receive no notification that a call is on hold when a call is placed on exclusive hold.

Refer also to “Call Hold: System” - section 8.38.

8.37.2 Interactions

If the Timeout on Exclusive Hold expires while a call is on Exclusive Hold, the call attempts to recall the responsible user. If the responsible user is unavailable, the call routes to the responsible user's applicable backup entity. If the backup entity is unavailable, the call routes to the applicable coverage entity for the responsible user.

Note
If a music source is connected to the Plexus system (see chapter 4, "Installation"), Music-On-Hold is broadcast to calls on Exclusive Hold.

8.37.3 Operation

8.37.3.1 Place a call on Call Hold: Exclusive

A. From a Plexus key telephone:

- Press the line button of the active call (the solid green light) once.
- It will then begin to flash slowly, indicating that the call is now on Call Hold: Exclusive.

or

- Press **FLASH** + **F02**

B. From an ordinary phone:

- Press **Flash** + **F02**

8.37.3.2 Retrieve a call placed on Call Hold: Exclusive

A. From a Plexus key telephone:

- Press the line button of the held call (the slow flashing green light).

or

- enter **F03**

B. From an ordinary phone:

- enter **Flash** + **F03**

or

- enter **Flash**

8.37.3.3 Retrieve a call placed on Call Hold: Exclusive from another extension

A. From a Plexus key telephone or an ordinary phone

- enter **F05** = User Id of user that placed the call on exclusive hold

8.37.4 Configuration

Table 8 - 28 Call Hold: Exclusive

Method	Path	Parameter	Default
Software	User: Features, Timeouts	Timeout on Exclusive Hold	5 minutes
Key Telephone	User: User Attributes	X-hold Timeout	5 minutes

8.37.5 Administration

N.A.

8.38 Call Hold: System

8.38.1 Description

System Hold is a non-exclusive hold that enables held calls to be retrieved by any user on the system. Calls placed on System Hold can be tagged for the target user with their User ID. Users retrieve tagged calls at any station by entering a feature code and their user ID. Held calls that are not tagged may be retrieved by entering a feature code and the associated trunk ID (or user ID if the held party is a system user) or by pressing the corresponding red flashing **DTS** or **DSS** key on a Plexus Key Telephone. Refer also to “Call Retrieve” - section 8.50.

8.38.2 Interactions

System held calls recall the user that placed them on hold (i.e., the responsible user) if the call is not retrieved before the System Hold Period expires. If the responsible user is unavailable, the call routes to the responsible user's applicable backup entity. If the backup entity is unavailable, the call routes to the applicable coverage entity for the responsible user.

Note
If a music source is connected to the Plexus system (see chapter 4, "Installation"), Music-On-Hold is broadcast to calls on System Hold.

8.38.3 Operation

A. From a Plexus key telephone:

Table 8 - 29 Call Hold: System

To perform this action	Do this
Place a call on Call Hold: System	Press **Hold.**
Place a call on System Hold with the recipient’s ID	**Flash** + **F01** + Recipient’s ID.

B. From an ordinary phone:

Table 8 - 30 Call Hold: System

To perform this action	Do this
Place a call on Call Hold: System	**Flash** + **F00**
Place a call on System Hold with the recipient’s ID	**Flash** + **F01** + Recipient’s ID.

8.38.4 Configuration

Table 8 - 31 Call Hold: System

Method	Path	Parameter	Default
Software	System Parameters: Features	System Hold Timeout	15 minutes
Key Telephone	N.A.		

8.38.5 Administration

It is recommended that a portion of the available programmable keys on Plexus Key Telephones be used for mapping trunks and users. Programmable keys defined as such are referred to as DTS (Direct Trunk Select) or DSS (Direct Station Select) keys and may be used to quickly retrieve held calls.

8.39 Call Intrude

8.39.1 Description

Call Intrude allows a user, with the appropriate privileges, to intrude on a busy line or user. Successful intrusions conference the intruding user with the active call. Call Intrude is available on internal, external, and conference calls.

8.39.2 Interactions

Calls on any type of hold (i.e., Consult, Exclusive, System) cannot be intruded on.

When Plexus Key Telephone users attempt to connect to a busy line or user for which there are intrude rights, the LED on the **OVERRIDE** button will flash.

Plexus Key Telephone users can grant all other users the right to intrude, on a per conversation basis, by pressing **OVERRIDE** during a call. The associated LED will appear solid red and all other users (even those without intrude privileges) will be able to intrude on the call by calling the user who has enabled the intrusion.

The displays on any Plexus Key Telephones will indicate 'Conferenced' when a user intrudes on an active call.

When a user intrudes on a call, all parties to the call will hear a notification tone (i.e., stutter dial tone).

8.39.3 Operation

A. From a Plexus key telephone:

- After receiving a busy signal, press **OVERRIDE**.

or

- After receiving a busy signal, press DSS key mapped with **Feature 12**

or

- After receiving a busy signal, press **FLASH** + **F12**

B. From an ordinary phone:

- After receiving a busy signal, **FLASH** + **F12.**

8.39.4 Configuration

The Intrude ALL privilege grants a user the right to intrude on all users and trunks. Limited intrude privileges may be granted by designating specific users on the user's intrude list instead of granting the universal privilege.

Table 8 - 32 Call Intrude

Method	Path	Parameter	Default
Software	User: Privileges User: Override, Intrude	Intrude ALL Name/ User ID	Not Granted None
Key Telephone	User: User Privileges	Can Intrude Call	No

8.39.5 Administration

N.A.

8.40 Call Monitor

8.40.1 Description

Call Monitor allows a user, with the appropriate privileges, to monitor or eavesdrop on a busy line or user. The monitoring user will be able to listen to, but will not be heard by, the parties on the call. However, when the user begins to monitor, a notification tone will be issued to all parties to indicate that the call is being monitored. Call Monitor is available on internal, external, and conference calls. Refer also to “Silent Monitoring” - section 8.119.

8.40.2 Interactions

Calls on any type of hold (i.e., Consult, Exclusive, System) cannot be monitored.

When a user begins to monitor a call, all parties to the call will hear a notification tone (i.e., stutter dial tone) and displays on any Plexus Key Telephones will indicate “Conferenced".

8.40.3 Operation

A. From a Plexus key telephone:

- After receiving a busy signal, press DSS key mapped with **Feature 13**

or

- After receiving a busy signal, press **FLASH** + **F13**

B. From an ordinary phone:

- After receiving a busy signal**, FLASH** + **F13**.

8.40.4 Configuration

The Monitor ALL privilege grants a user the right to monitor all users and trunks. Limited monitor privileges may be granted by designating specific users on the user’s monitor list instead of granting the universal privilege.

Table 8 - 33 Call Monitor

Method	Path	Parameter	Default
Software	User: Privileges User: Override, Monitor	Monitor ALL Name/ User ID	Not Granted None
Key Telephone	User: Privileges	Can Monitor Call	No

8.40.5 Administration

N.A.

8.41 Call Notify

8.41.1 Description

Call Notify enables a user to be notified of calls ringing to monitored users, calls ringing to monitored trunks, and calls in the Hold Queue of a monitored user group. The system will notify the user by sending a stutter tone through the user's speaker or handset.

Note
This feature is not available on version 1.00 and 1.01 Keyphones or ordinary telephones.

8.41.2 Interactions

On 30 button Keyphone, DSS Key 13 must be programmed as Utility: Call Notify.

On 22 button Keyphone, DSS Key 04 must be programmed as Utility: Call Notify.

A solid red light indicates Call Notify enabled. No light indicates Call Notify disabled. A user can also enable or disable the Call Notify in the programming mode of the Keyphone as a keyphone option.

To be notified of calls ringing to monitored users, monitored user groups, and monitored trunks, the user must have Universal Answer privilege enabled. This privilege allows the user to pickup these calls by pressing the associated flashing **DSS** or **DTS** key**.**

To be notified of calls in the Hold Queue of a monitored user group, the user must have Universal Retrieve privilege enabled. This privilege allows the user to pickup the call in the Hold Queue by pressing the associated flashing **DSS** key.

Note
Call Notify will only work with mapped keys on a Keyphone, not a DSS-60/ DDSS-60 Console. An operator with a DSS-60/DDSS-60 Console must map users, user groups, and trunks on their Keyphone that they wish to have Call Notify track.

8.41.3 Operation

- The user presses the **Call Notify** key (see "Configuration" below) to be notified of calls. The light will be lit red if Call Notify is enabled. The light will not be lit if Call Notify is not lit.
- The Key Telephone will notify the user of a ringing call or a call in a user group Hold Queue.
- If the user wished to answer or retrieve the call, the user presses the associated flashing **DSS** or **DTS** key.

8.41.4 Configuration

On 30 button Keyphone, DSS Key 13 must be programmed as Utility: Call Notify.

On 22 button Keyphone, DSS Key 04 must be programmed as Utility: Call Notify.

A user can also enable or disable the Call Notify in the programming mode of the Keyphone as a keyphone option.

Table 8 - 34 Call Notify

Method	Path	Parameter	Default
Software	User: Privileges	Univ. Answer Univ. Retrieve	Not Granted Not Granted
Key Telephone	User: User Privileges	Can Universal Answer Can Universal Retrieve	No No

8.41.5 Administration

N.A.

8.42 Call Park

8.42.1 Description

The Call Park feature addresses situations where a user encounters a busy user in the process of transferring a call. Call Park enables a user to transfer either an internal or external call into a target user's Busy Hold queue so that the transfer can be completed. The busy target user will hear a stutter tone as an indication that a call has entered their queue.

8.42.2 Interactions

Once the caller has been parked in the queue, the call will be processed as according to the user's settings for Busy Queuing - User (see "Busy Queuing - User" - section 8.21).

8.42.3 Operation

8.42.3.1 Manual Parking

A. From a Plexus key telephone:

- With the caller on the line, press **XFER.**
- Dial the desired user ID/ user group ID or press a mapped **DSS** key.
- After receiving a busy signal, press **CONF**, and hang up (go on-hook).

B. From an ordinary phone:

- With the caller on the line, **FLASH**.
- Dial the desired User ID.
- After receiving a busy signal, **FLASH** + **F17**.
- Hang up (go on-hook).

8.42.3.2 Automatic Parking

A. From a Plexus key telephone:

- With the caller on the line, press **XFER.**

- Dial the desired User ID.
- After receiving a busy signal, hang up (go on-hook).

B. From an ordinary phone:

- With the caller on the line, **FLASH**.
- Dial the desired User ID.
- After receiving a busy signal, hang up (go on-hook).

Note
The target user must have "Busy Queuing - User" enabled for automatic parking to work properly (see “Busy Queuing - User” - section 8.21).

8.42.4 Configuration

In order for a user to automatically park a call for a busy user, the busy user must have Busy Queuing - User enabled. If the busy target user does not have Busy Queuing - User enabled, calls can be manually parked using the Camp-On feature described above. Otherwise, parked calls route to the target user’s busy coverage.

See“Busy Queuing - User” - section 8.21.

8.42.5 Administration

N.A.

8.43 Call Pickup: Inter-Group

8.43.1 Description

Inter-Group Call Pickup enables a user to answer the ringing station of a user in a pickup group other then their assigned pickup group. The user must specify the user group ID of the desired pickup group. If multiple stations in the pickup group are ringing, the system selects one of the calls. Refer also to “Call Pickup: Intra-Group” - section 8.44 and “Universal Answer” - section 8.134.

Note
Users need not be assigned a pickup group to use the Inter-Group Call Pickup feature. However, the user must have the Universal Answer Privilege enabled to use this feature

8.43.2 Interactions

If no station in the specified pickup group is ringing when this feature is utilized, the user will hear error tone. The user must have the Universal Answer Privilege enabled to use this feature.

8.43.3 Operation

A. From either a Plexus key telephone or an ordinary phone**:**

- To answer a ringing station in another pickup group, enter **F44** + User Group ID.

8.43.4 Configuration

Table 8 - 35 Call Pickup: Inter-Group

Method	Path	Parameter	Default
Software	User: Privileges	Universal Answer	Not Granted
Key Telephone	User: Privileges	Can Universal Answer	No

8.43.5 Administration

N.A.

8.44 Call Pickup: Intra-Group

8.44.1 Description

Intra-Group Call Pickup enables a user to answer the ringing station of any user in their assigned pickup group. If multiple stations in the same pickup group are ringing, a user can either allow the system to select one of the calls or specify a particular call by entering the associated user ID. Refer also to “Call Pickup: Inter-Group” - section 8.43 and “Universal Answer” - section 8.134.

8.44.2 Interactions

If no station in the user's assigned pickup group is ringing when this feature is utilized, the user will hear error tone.

8.44.3 Operation

A. From either a Plexus key telephone or an ordinary phone:

Table 8 - 36 Call Pickup: Intra-Group

To perform this action	Do this
Answer any ringing station within the pickup group	Enter **F07**.
Answer a specific ringing station within the pickup group	Enter **F08** + User ID.

8.44.4 Configuration

Table 8 - 37 Call Pickup: Intra-Group

Method	Path	Parameter	Default
Software	User: Assignment, Group Access	Pilot-ID for Pickup	None
Key Telephone	User: User Attributes	Pickup Pilot ID	0

8.44.5 Administration

Additional user groups may be created to further expand the utility of the call pickup feature. For example, if Joe in Sales needs to be able to answer Sheila's calls as well as the calls of the Sales group, a new group may be created that includes all of the members of the Sales group plus Sheila.

8.45 Call Progress Tones

8.45.1 Description

Plexus systems are capable of detecting and generating several Call Progress tones. Tone detection is enabled by installing and activating an appropriate filter. The frequency, level, and cadence of tones generated by the system is controlled through Plexus Administrator. Refer also to “CO Disconnect Supervision” - section 8.23 and “Fax Detection” - section 8.73.

8.45.2 Interactions

Up to three filters may be installed. Installed filters are activated and used for detecting certain types of tones.

8.45.3 Operation

N.A.

8.45.4 Configuration

The only filter installed by default is a filter for CNG tone used for the Fax Detection feature. Additional filters for other types of tones may be requested by writing to bbstechs@bbstelecom.com. (Specify the frequency range you need.)

There are several configurable parameters that may be utilized to further define the characteristics of each activated filter.

The characteristics of system generated tones are modified in the Generator section.

Table 8 - 38 Call Progress Tones

Method	Path	Parameter
Software	System Parameters: Call Progress, Filter	Installed filters appear here. Filters are activated for the applicable tone.
	System Parameters: Call Progress, Detector	The following parameters are configured. Cadenced min. Cycles min.ON Time
	System Parameters: Call Progress, Generator	max.ON Time min.OFF Time max.OFF Time The frequency, level, and cadence are selected for Ringback, Busy, Error, and Dial tone. The Amplitude is selected for the low and high frequencies associated with DTMF tones.

Note
These parameters cannot be modified through the keyphone.

8.45.5 Administration

N.A.

8.46 Call Priority

8.46.1 Description

Plexus systems make use of a feature called Call Priority to determine which of two or more simultaneous calls is completed to the target user. The system prioritizes calls based on the weighted value of each of several characteristics that a call may possess. The call characteristics and their weighted values are as follows:

Table 8 - 39 Call Priority

Characteristic	Value
Internal	0
Hands-Free	1
Forwarded	2
Covered*	4
Transferred	8
Call Back	16
Recall	32
Direct	64
External	128

* A covered call is a call transferred by the system according to a call coverage setting.

To determine the weighted value of a call, add the values for each characteristic applicable to the call. The simultaneous call with the greatest weighted value is given priority over the other calls and is completed to the target user. The other call(s) routes to the target user's busy coverage.

Note
External calls are always a higher priority than internal calls.

8.46.1.1 Example #1

An external call transferred by an auto-attendant vs. a direct external call, as in the case of a trunk directly routed to an extension on the system.

Table 8 - 40 Call Priority

Auto-Attendant		Direct	
External	128	External	128
Transferred	8	Direct	64
Total	136	Total	192

The direct external call has a greater weighted value and is therefore completed to the intended user.

8.46.1.2 Example #2

A direct internal call vs. a covered internal call.

Table 8 - 41 Call Priority

Direct		Covered	
Internal	0	Internal	0
Direct	64	Covered	4
Total	64	Total	4

The direct internal call has a greater weighted value and is therefore completed to the target user.

8.46.2 Interactions

N.A.

8.46.3 Operation

N.A.

8.46.4 Configuration

N.A.

8.46.5 Administration

N.A.

8.47 Call Recall

8.47.1 Description

Call Recall is a system feature which causes a held call to recall the user which placed the call on hold (i.e., the responsible user) upon expiration of a programmed timeout period. This ensures that calls placed on hold are not forgotten.

8.47.2 Interactions

Call Recall interacts with the responsible user's Backup and Coverage. Please refer to the following sections for details: "Call Hold: Consult" - section 8.36, "Call Hold: Exclusive" - section 8.37, and "Call Hold: System" - section 8.38.

8.47.3 Operation

N.A.

8.47.4 Configuration

N.A.

8.47.5 Administration

N.A.

8.48 Call Record

8.48.1 Description

The Call Record function enables Plexus Key Telephone users to record telephone conversations in real-time. The user need only press an appropriate programmable key defined as **Function 1** . The recorded conversation is automatically saved as a message in the user's assigned voice mailbox when the user hangs up.

Note
Up to 195 minutes of a call can be recorded and stored as a single message in the user's mailbox.

8.48.2 Interactions

N.A.

8.48.3 Operation

A. From a Plexus key telephone:

- While on a call, press the **RECORD** key.
- A click will be heard indicating that recording has begun.
- To finish recording, press the **RECORD** key or hang up.
- When the user hangs up (goes on-hook), the recording will be stored in their mailbox.

8.48.4 Configuration

The Integrated Voice Processor (IVP) must be configured correctly for this function to work. Refer to chapter 9, "Integrated Voice Processor" for more information.

8.48.5 Administration

The Plexus Key Telephone must have one of the programmable keys defined with the Call Record function code (**Function 1**).Call Record is a function of the Plexus Keyphone. It is not possible to use this function with ordinary (single line) telephones.

8.49 Call Release

8.49.1 Description

Call Release allows the user to selectively drop (release) individual parties from a call conference.

Refer to Call Conference for more information about call conferencing.

8.49.2 Interactions

The system recognizes each party in a call conference as a single call. For example, if a user presses either a Call Broker key or a DTS key associated with a conference party, all of the parties to the conference will be placed on Exclusive hold. When the Release key is pressed on the Plexus Key Telephone, each party in the call conference is assigned a separate key on the phone. Individuals may then be dropped from the call conference by pressing the key to which they have been assigned.

8.49.3 Operation

A. From a Plexus key telephone:

- While in a call conference, press the **RELEASE** key.
- Then, press the **DSS** key of the party to be dropped.

8.49.4 Configuration

No configuration is necessary; the **RELEASE** key is a feature of the Plexus key telephone.

8.49.5 Administration

N.A.

8.50 Call Retrieve

8.50.1 Description

Call Retrieve is used to retrieve calls placed on either Exclusive Hold or System Hold. Several forms of Call Retrieve are available as follows:

8.50.1.1 Personally Held

Retrieves any type of held call to the responsible user (i.e., any call placed on Exclusive Hold).

8.50.1.2 System Hold-Holding Party Specified

Retrieves a specific call on System Hold by entering a feature code and the trunk ID of the held call (or user ID if the held party is a system user).

8.50.1.3 System Hold-Intended User Specified

Retrieves a tagged call on System Hold by entering a feature code and the intended user's user ID.

8.50.1.4 System Hold-FIFO

Retrieves any call on System Hold according to a First In First Out (FIFO) convention.

Refer also to "Call Hold: Exclusive" - section 8.37 and "Call Hold: System" - section 8.38.

8.50.2 Interactions

While the various forms of Call Retrieve are available to all system users they are most valuable to ordinary telephone users. In most cases, Plexus Key Telephone users will retrieve held calls by pressing the corresponding **DTS** or **DSS** key.

8.50.3 Operation

A. From a Plexus key telephone:

- Exclusive:Press **F03**.
- Holding Party:Press **F04** + User ID.
- Intended User:Press **F05** + User ID.
- FIFO:Press **F06**.

8.50.4 Configuration

N.A.

8.50.5 Administration

It is recommended that a portion of the available programmable keys on Plexus Key Telephones be used for mapping trunks and users. Programmable keys defined as such are referred to as **DTS** (Direct Trunk Select) or **DSS** (Direct Station Select) keys and may be used to quickly retrieve held calls.

8.51 Call Transfer: Screened

Also known as Supervised Transfer. This is only for calls transferred by the Integrated Voice Processor's Auto Attendant.

8.51.1 Description

Screened Call Transfer enables users to screen calls transferred by the Integrated Voice Processor's Auto-Attendant. When properly configured, the Auto-Attendant can request and record a caller's name and then play the recording to the intended user before completing the transfer. After listening to the recording the intended user can accept the call, listen to the recording again, route the call to their voice mailbox, or place the call on Exclusive Hold.

8.51.2 Interactions

N.A.

8.51.3 Operation

N.A.

8.51.4 Configuration

The Integrated Voice Processor (IVP) must be configured correctly for this feature to work. Refer to chapter 9, "Integrated Voice Processor" for more information.

8.51.5 Administration

N.A.

8.52 Call Transfer: Supervised

8.52.1 Description

Call Transfer allows a user to transfer either an internal or external call to another user (or trunk). A Supervised Call Transfer occurs when the transferring user speaks with the target user before completing the transfer. With Supervised Call Transfer, any number of attempts can be made to locate the target user without terminating the call.

8.52.2 Interactions

N.A.

8.52.3 Operation

A. From a Plexus key telephone:

8.52.3.1 If the target user is mapped on a programmable **DSS** Key:

- Press the mapped **DSS** key of the target user
- Announce the call if desired.
- Hang up (go on-hook).
- If the user is busy, not available, or DND, press the flashing **DSS** key associated with the held call to retrieve the call and select another user or press **MSG** and hang up to forward the call to the target user's backup/ coverage.

or

8.52.3.2 If the target user is not mapped on a programmable **DSS** key:

- Press **XFER**.
- Dial the User ID.
- Announce the call if desired.
- Hang up (go on-hook).

- If the user is busy, not available, or DND, press the flashing **DSS** key associated with the held call to retrieve the call and select another user or press **MSG** and hang up to forward the call to the target user's backup/ coverage.

B. From an ordinary phone:

- **FLASH**.
- Dial the User ID.
- Announce the call if desired.
- Hang up (go on-hook).
- If the user is busy, not available, or DND, **FLASH**, and select another user.

8.52.4 Configuration

In order for a user to transfer one external caller to another external caller (i.e., conference or connect two external calls), the Unattend Trunks privilege must be granted.

Table 8 - 42 Call Transfer: Supervised

Method	Path	Parameter	Default
Software	User: Privileges	Unattend Trunks	Disabled
Key Telephone	User: User Privileges	Can Unattend Trunks	No

8.52.5 Administration

It is recommended that a portion of the available programmable keys on Plexus Key Telephones be used for mapping trunks and users. Programmable keys defined as such are referred to as DTS (Direct Trunk Select) or DSS (Direct Station Select) keys and may be used to quickly transfer a call to the associated user or trunk.

8.53 Call Transfer: Unsupervised

Also known as Blind Transfer

8.53.1 Description

Call Transfer allows a user to transfer either an internal or external call to another user (or trunk). An Unsupervised Call Transfer occurs when the transferring user completes the transfer without first speaking to the target user. Refer also to “Call Transfer: Supervised” - section 8.52.

8.53.2 Interactions

If the target user is unavailable, the call routes to the target user’s backup or coverage (see “Transfer Recall” - section 8.131 for exceptions).

If the target user is busy and has Busy Queuing - User enabled, the call will automatically be placed in the target user’s Exclusive Hold queue (see “Busy Queuing - User” - section 8.21).

8.53.3 Operation

A. From a Plexus key telephone:

8.53.3.1 If the target user is mapped on a programmable **DSS** Key:

- Press the mapped **DSS** key of the target user
- Announce the call if desired.
- Hang up (go on-hook).
- If the user is busy, not available, or DND, press the flashing **DSS** key associated with the held call to retrieve the call and select another user or press **MSG** and hang up to forward the call to the target user’s backup/ coverage.

or

8.53.3.2 If the target user is not mapped on a programmable DSS key:

- Press **XFER**.
- Dial the User ID.

- Hang up (go on-hook).
- If the user is busy, not available, or DND, press the flashing **DSS** key associated with the held call to retrieve the call and select another user or press **MSG** and hang up to forward the call to the target user's backup/ coverage.

B. From an ordinary phone:

- Press **FLASH.**
- Dial the user ID.
- Hang up.

8.53.4 Configuration

In order for a user to transfer one external caller to another external caller (i.e., conference or connect two external calls), the Unattend Trunks privilege must be granted.

Table 8 - 43 Call Transfer: Unsupervised

Method	Path	Parameter	Default
Software	User: Privileges	Unattend Trunks	Disabled
Key Telephone	User: User Privileges	Can Unattend Trunks	No

8.53.5 Administration

N.A.

8.54 Call Waiting

8.54.1 Description

The Call Waiting feature, when activated, provides a stutter tone to busy users to indicate that a call is waiting. The busy user can choose to answer the waiting call or allow it to route to No Answer coverage.

Note
Call Waiting is not available for calls routed to a user group. Such calls always route according to the defined distribution method (see “Call Distribution Management” - section 8.33). Refer also to “Busy Queuing - User Group” - section 8.22 and “Intercept Service” - section 8.84.

8.54.2 Interactions

Waiting calls continue to ring the busy user until the Timeout on No Answer expires. The call then routes to the No Answer coverage for the busy user.

The CALL WAIT LED on a Plexus Key Telephone appears solid red when Call Waiting is activated and flashes to indicate a waiting call.

Caller ID information for waiting calls appears on the display of a Plexus Key Telephone at the time the associated call is received.

Activating Call Waiting on an ordinary telephone interferes with the ability to use the Call Conference feature at the associated station.

8.54.2.1 Operation

A. From a Plexus key telephone:

Table 8 - 44 Call Waiting

To perform this action	Do this
Activate/deactivate Call Waiting mode	Press **Call Wait**.
Accept a waiting call	Press **Call Wait** or press the flashing D**SS/Call** key to answer a specific call
Reject a waiting call	Press **DND.**

Note
If a user rejects an incoming call by pressing the **DND** key, the call will be routed to the user's busy backup / coverage entities (see "Call Backup - User / User Group" - section 8.27 and "Call Coverage" - section 8.32).

B. From an ordinary phone:

Table 8 - 45 Call Waiting

To perform this action	Do this
Activate Call Waiting mode	Press **F71**.
Deactivate Call Waiting mode	Press **F70**.
Accept a waiting call	**Flash**. The current call will be placed on Call Hold: Consult.
Return to the current call	**Flash** again.

8.54.3 Configuration

Call Waiting is activated and deactivated by the user. Both Plexus Key Telephone and ordinary telephone stations will not have this feature enabled by default.

8.54.4 Administration

N.A.

8.55 Caller ID

Plexus systems support the display and storage of Caller ID information at stations with a Plexus Key Telephone (display model only). Caller ID information (when available) appears on the display of the Plexus Key Telephone at the time the associated call is received. The available Caller ID information for each call is stored for later review by the user.

8.55.1 Interactions

The following messages appear in cases where the Caller ID information is not available or properly received.

```
UNAVAILABLE
WITHHELD
OUT OF AREA
CORRUPTED
```

Caller ID information does not display on a Supervised Call Transfer until the transfer is completed by the transferring user.

Caller ID information only displays on Plexus Key Telephones (display model only). It does not display on ordinary Caller ID devices connected to Plexus extension ports.

Plexus Key Telephone users (display model only) can view available Caller ID information for any active, external call on system hold.

Plexus Key Telephone users (display model only) can view call information associated with any busy user or in-use line (see “Busy Inquiry” - section 8.20). Call information does not include the Caller ID.

If Caller ID is not enabled, the display will always indicate that call information is “Unavailable”.

If Caller ID is enabled, but no service is provided by the CO, the display will always indicate that call information is “Unavailable”.

Caller ID information (when available) is included in SMDR records (see “Station Message Detail Recording (SMDR)” - section 8.124).

8.55.2 Operation

8.55.2.1 Changing the Caller ID display format

A. From a Plexus key telephone:

- Press and hold down **OVERRIDE** for 2 seconds.
- Press **VOL ▼** until the option **Set CallerId Fmt** appears.
- Press **SPKR** to select this mode.
- Use **VOL ▼** and **VOL ▲** to position the dash.
- Use **CONF** and **XFER** to position the parenthesis.
- Press **OVERRIDE** to accept and exit

8.55.2.2 Viewing Caller ID entries

A. From a Plexus key telephone:

- Press the **CALLER ID** key.
- The **SCREEN** key may be used to scroll through the entries.

8.55.2.3 Deleting Caller ID entries

A. From a Plexus key telephone:

- Press the **FLASH** key to delete entries.

8.55.3 Configuration

Caller ID service must be activated on the telephone lines in order to receive Caller ID information. Contact the local telephone company.

Only the Caller ID version of the Analog Trunk Interface (ATI) is capable of receiving and processing Caller ID information.

Caller ID must be enabled on each trunk.

Table 8 - 46 Caller ID

Method	Path	Parameter	Default
Software	Trunk Port: Features	Caller-ID Enabled	Disabled
Key Telephone	N.A.		

8.55.4 Administration

The display format of the Caller ID information, including the position of the parentheses and the hyphen, is configurable by the user.

8.56 Calling Party Display: Internal

Internal calls placed and received on Plexus Key Telephones (display model only) display the user ID and user name of the target user or calling party; similar to Caller ID on incoming external calls.

8.56.1 Interactions

N.A.

8.56.2 Operation

N.A.

8.56.3 Configuration

N.A.

8.56.4 Administration

N.A.

8.57 Caller ID Redial

8.57.1 Description

Allows users to returns calls using the Caller ID info in the Keyphone Caller ID buffers.

8.57.2 Interactions

The user presses '**Caller ID**' key and scrolls (using '**screen**' key) to the desired number. The user presses **redial**. The user presses '**1**' for local call (no 1 or area code dialed), '**2**' for regional call (area code dialed), '**3**' for national call (national call prefix + area code dialed), and '**4**' for international calls (international call prefix + country code dialed). System: DXP: Feature: Limits: Dialing Prefix-Code for: International/National/Area Width must be defined.

Note
Caller ID Redial is not available with version 1.00 and 1.01 Plexus Keyphones and Ordinary Phones.

8.57.3 Operation

A. From a Plexus key telephone:

- Press **CALLER ID** key.
- Press **SCREEN** to scroll through the Caller ID buffer to the desired call back number.
- Press **REDIAL** key.
- Press **1** for local calls, **2** for regional calls, **3** for National calls, or **4** for international calls.

8.57.4 Configuration

Table 8 - 47 Caller ID Redial

Method	Path	Parameter	Default
Software	System Parameter: Feature: Limits: Dial Prefix Code for	International National Area Width	011 1 3
Key Telephone	N.A.		

Note
If a user attempts to call back a number that falls in the user's Deny Dialing table, the call will fail (see "Toll Restriction" - section 8.129).

8.57.5 Administration

N.A.

8.58 Camp-On

8.58.1 Description

The Camp-On feature addresses situations where a user encounters a busy response when calling another user. Camp-On enables a user to place themselves into a busy target user's Exclusive Hold queue. The busy user will hear a stutter tone as an indication that a call has entered their queue.

8.58.2 Interactions

Once the calling user is in the busy user's queue, all Exclusive Hold interactions apply as if the busy user placed the calling user on Exclusive Hold themselves (see "Call Hold: Exclusive" - section 8.37).

Camp-On may be used to manually park external calls in cases where a busy user does not have Call Queuing enabled (see "Busy Queuing - User" - section 8.21). This is accomplished by camping on with the call and then hanging up. The call will remain in the busy user's Exclusive Hold queue. See "Call Park" - section 8.42.

8.58.3 Operation

A. From a Plexus key telephone:

- After receiving a busy signal, press **CONF**.

B. From an ordinary phone:

- After receiving a busy signal**, FLASH** + **F15**.

8.58.4 Configuration

N.A.

8.58.5 Administration

8.59 Centrex compatibility

8.59.1 Description

Centrex is a collection of business telephone services offered by local telephone companies. Centrex features, such as three-party conferencing and off-site transferring, are often accessed by sending a 'flash' signal to the telephone company central office (CO). Because Plexus systems are capable of sending 'flash' signals to the CO, they are compatible with most Centrex services.

If an external call entered on a Centrex enabled Trunk and is being routed by the system to a 2nd External number, the system will hook flash the Trunk, place the forwarding call, and hook flash again to release the call, causing the call to be transferred through the Centrex Trunk.

Refer also to "Flash: Central Office" - section 8.74.

8.59.2 Interactions

With Centrex services, it is possible to place calls on hold at the CO. While users can utilize this Centrex feature, the Plexus system has no means of managing such held calls.

8.59.3 Operation

8.59.3.1 For a call received by a user:

A. From a Plexus key telephone:

- Press **FLASH** + **F54**.

B. From an ordinary phone:

- To send a 'flash' signal to the central office, **FLASH** + **F54**.

8.59.3.2 For a call forwarded by the system:

Note
Centrex forwarding by the Plexus system will not be enabled for initial release of Plexus Administrator version 2.0. Contact your representative for information on implementation dates.

No setup or operation is necessary. If a call comes in on a Centrex enabled line and is being externally forwarded by the system, the system will automatically forward the call out on the same line, using Centrex flashing.

8.59.4 Configuration

8.59.4.1 For a call received by a user:

The ability to send a 'flash' signal to the CO is configured on a user-basis.

When activating Centrex services with the local telephone company, inquire about the required 'flash' signal duration. To assure compatibility, adjust the CO Flash Period accordingly.

Table 8 - 48 Centrex Compatibility

Method	Path	Parameter	Default
Software	User: Privileges Trunk Port: Features	CO Flash CO Flash Period	Disabled 480 ms
Key Telephone	User: Privileges	Can Flash CO	No

8.59.4.2 For a call forwarded by the system:

Table 8 - 49 Centrex Compatibility

Method	Path	Parameter	Default
Software	Trunk Port: Features	Centrex Enabled	Disabled
Key Telephone	N.A.		

8.59.5 Administration

N.A.

8.60 Class of Service: User

8.60.1 Description

Configuration of Plexus systems involves the definition of a Class of Service for each user. Several programmable parameters determine which features and privileges are available to each user and how the user's calls are routed, queued, and covered. A suitable Class of Service is developed for a user based on their job function, responsibilities, authority, security clearance, equipment (i.e., ordinary phone or key telephone), and preferences.

Note
Plexus Administrator enables parameter definitions and assignments to be replicated to all or selected users. This feature is useful in cases where several users will share a similar Class of Service.

8.60.2 Interactions

N.A.

8.60.3 Operation

N.A.

8.60.4 Configuration

A Class of Service is defined by addressing all of the user-specific parameters through either Plexus Administrator or Key Telephone Configuration.

8.60.5 Administration

N.A.

8.61 CTi Interface

See chapter 16, "CTi Interface".

8.62 Day/Night Service

8.62.1 Description

Day/Night Service enables Plexus systems to route incoming calls differently based on the day and time (Day or Night) a call is received. For example, during the day, the system may be programmed to route incoming calls to live operators, while at night incoming calls may be routed to an Auto-Attendant.

8.62.2 Interactions

The Day/Night Service is used in routing incoming calls. Incoming calls can be routed to system resources (typically users, user groups, auto-attendants, or voice mail greetings) or to an external number (e.g., answering service or remote user).

The transitions between Day/Night are handled by the system's real-time clock. The resolution of the real-time clock is +/- 1 minute. It is possible to manually force the system to switch between Day/Night mode.

The Day/Night service is also used to define the times for validating trunk access by users (see "Authorize User" - section 8.12).

8.62.3 Operation

8.62.3.1 Manually switching between Day and Night mode

A. From a Plexus key telephone:

- Define a programmable **DSS/DTS** key as **Function 4 - Day/Night** key (the user must have **Administrate** privileges).
- Press the **Day/Night** key to switch time modes. A solid red light indicates actual Night Mode. A Blinking red light indicates forced Night Mode. A solid green light indicates actual Day Mode. A blinking green light indicates forced Day Mode.

B. From either a Plexus key telephone or an ordinary phone:

- Activate System Administrator Privileges.
- Enter **F48** for forced Day mode.
- Enter **F49** for forced Night mode.
- Enter **F47** to return to automatic settings/actual time.

Note
If the system is placed in forced Day or forced Night mode, the system will not switch modes automatically until being returned to automatic settings/actual time.

8.62.4 Configuration

To ensure precise Day/Night transitions, the date and time must be set correctly. The system clock is set using the Update: Clock feature of Plexus Administrator (see chapter 6, "Software Configuration") or through Key Telephone Configuration. Day and night routing assignments must be made for each trunk.

Table 8 - 50 Day/Night Service

Method	Path	Parameter	Default
Software	System Parameters: Time: Mode: Day of Week: Sunday thru Saturday	Mode Day Time Night Time	Automatic 8:00 AM 6:00 PM
	Trunk: Routing: Voice: Day	Routing Method Routing Entity	Fixed None
	Trunk: Routing: Voice: Night	Routing Method Routing Entity	Fixed None
Key Telephone	System Time: Day/Night Time: Sunday thru Saturday	Mode Daytime Hour Daytime Minute Nighttime Hour Nighttime Minute	Automatic 8 AM 00 6 PM 00
	Trunks: Trunk Number	Day Routing Type Day Routing Day Routing Dial # Night Routing Type Night Routing Night Routing Dial #	Fixed 0 0 Fixed 0 0

8.62.5 Administration

N.A.

8.63 Direct Inward System Access (DISA)

8.63.1 Description

The Direct Inward System Access (DISA) feature allows individuals to dial into the system and directly access any resource (e.g., user, user group, trunk, trunk group) on the system. DISA works much like an auto-attendant. However, with DISA it is not necessary to listen to a greeting before directing the call. DISA functionality goes beyond that of an auto-attendant in that the individual can directly access a trunk to make an outbound call through the system. This allows individuals to make business-related long distance calls from a remote site using the company's long distance service. Refer also to "Remote Station Service (RSS)" - section 8.116.

8.63.2 Interactions

N.A.

8.63.3 Operation

In order to utilize DISA, a user must call into a DISA-enabled trunk, enter the DISA access digit within the defined Activation Timeout period, and then enter the correct DISA Password.

8.63.4 Configuration

DISA configuration involves enabling the feature and defining the Activation Timeout and the Session Timeout. The Activation Timeout is the period of time that a caller is given to enter the DISA access digit. The Session Timeout is the time allotted for a DISA session. Upon expiration of the Session Timeout, DISA calls are automatically terminated by the system. The selected access digit and DISA password apply to all DISA-enabled trunks on the system.

Table 8 - 51 Direct Inward System Access (DISA)

Method	Path	Parameter	Default
Software	Trunk: Features	DISA Enabled	Disabled
		Activation Timeout	0 seconds
		Session Timeout	15 minutes
	System Parameters: General	DISA Password	123456
	System Parameters: Dial Plan	Access Digit for DISA	*
Key Telephone	Trunk: Trunk Number	DISA Enabled	No
		DISA In-Dial TO	0
		DISA Session TO	15

8.63.5 Administration

N.A.

8.64 Direct Station Select (DSS)/BLF

8.64.1 Description

Plexus Key Telephone users can assign programmable keys to other users on the system for the purpose of accessing and monitoring the user. The access function, known as Direct Station Select (DSS), enables the key telephone user to press the assigned programmable key to be connected to the corresponding user. The mapping function, made possible by LEDs (Light Emitting Diodes) on the programmable keys, is referred to as Busy Lamp Field (BLF). Different LED colors and states indicate a different status for the mapped user [See the **PVT-30D/PDT-30D Telephone User's Guide** for a complete listing of the various LED states]. For example, a solid red LED indicates that the corresponding user is currently on a call or an amber LED (2.0 Keyphones and above) indicates that the corresponding user is currently DND.

Note
DSS/BLF is also available on the Plexus Operator Console.

8.64.2 Interactions

N.A.

8.64.3 Operation

A. From a Plexus key telephone:

- Press a User ID **DSS** button to initiate a call to the corresponding user.

8.64.4 Configuration

The system administrator can use Plexus Administrator to assign users' keys. Users may assign their keys by entering programming mode on their key telephone

Table 8 - 52 Direct Station Select (DSS)/BLF

Method	Path	Parameter	Default
Software	User: Features, Settings	Define Key… on Key-Telephone	Unassigned
Key Telephone	See the **PVT-30D/PDT-30D Telephone User's Guide**		

8.64.5 Administration

See the **PVT-30D/PDT-30D Telephone User's Guide** for information on assigning DSS keys manually.

8.65 Direct Trunk Select (DTS)/BLF

Plexus Key Telephone users can assign programmable keys to outside lines (i.e., trunks) on the system for the purpose of accessing and mapping the trunk. The access function, known as Direct Trunk Select (DTS), enables the key telephone user to press the assigned programmable key to be connected to the corresponding trunk. The mapping function, made possible by LEDs (Light Emitting Diodes) on the programmable keys, is referred to as Busy Lamp Field (BLF). Different LED colors and states indicate a different status for the mapped trunk [See the **PVT-30D/PDT-30D Telephone User's Guide** for a complete listing of the various LED states]. For example, a flashing green LED indicates that the corresponding trunk is currently on Exclusive Hold for the user or (2.0 Keyphones or greater) was placed on hold from that Keyphone.

Note
DTS/BLF is also available on the Plexus Operator Console.

8.65.1 Interactions

N.A.

8.65.2 Operation

A. From a Plexus key telephone:

- Press a Trunk ID **DTS** button to initiate access to the corresponding trunk.

8.65.3 Configuration

The system administrator can use Plexus Administrator to assign users' keys. Users may assign their keys by entering programming mode on their key telephone.

Table 8 - 53 Direct Trunk Select (DTS)/BLF

Method	Path	Parameter	Default
Software	User: Features, Settings	Define Key… on Key-Telephone	Unassigned
Key Telephone	See the **PVT-30D/PDT-30D Telephone User's Guide**		

8.65.4 Administration

See the **PVT-30D/PDT-30D Telephone User's Guide** for information on assigning DTS keys manually.

8.66 Discriminating Ring

8.66.1 Description

Discriminating Ring enables different ringing patterns on ordinary phones in order to distinguish between internal and external calls and calls ringing at different stations within close proximity of one another. Refer also to “PBX Operation” - section 8.100.

8.66.2 Interactions

N.A.

8.66.3 Operation

N.A.

8.66.4 Configuration

Internal and external ring patterns are developed for each extension port where an ordinary phone is present by creating a cadence within the ON position of the ringback tone.

Table 8 - 54 Discriminating Ring

Method	Path	Parameter	Default
Software	Extension Port: SLT	Ring Pattern Int Ring Pattern Ext	60, 20, 20 Continuous
Key Telephone	N.A.		

8.66.5 Administration

The discriminating ring pattern is set by adjusting bitmap values (between 0 and 65535) which affect the modulation of the standard ring.

8.67 Distinctive Ring

Distinctive Ring allows Plexus Key Telephone users to select different internal and external ring types from sixteen available tones. The different ring types enable users to distinguish between internal and external calls and calls ringing at different stations within close proximity of one another.

8.67.1 Interactions

N.A.

8.67.2 Operation

A. From a Plexus key telephone:

- Press and hold down **OVERRIDE** for 2 seconds.
- Press **VOL ▼** until the option **Int. Ring Type** or **Ext. Ring Type** appears.
- Press **CONF** or **XFER** to choose a ring type.
- Press **OVERRIDE** to accept and exit.

8.67.3 Configuration

Internal and external ring types are selected by the user through programming mode on the Plexus Key Telephone.

See the **PVT-30D/PDT-30D Telephone User's Guide** for information on internal and external ring types.

8.67.4 Administration

N.A.

8.68 Do Not Disturb (DND)

8.68.1 Description

Do Not Disturb (DND) allows a user to prevent all internal and external from ringing at their station. Paging announcements are also disabled.

Note
With the proper privileges, a user can override another user's DND mode (see Do Not Disturb (DND) Override).

8.68.2 Interactions

Plexus Key Telephone users can enable DND mode after an incoming call begins to ring at their station. If the incoming call is an internal call and the calling party does not have Auto Cover enabled, the calling party will hear DND error tone. If the incoming call is an external call, the calling party will immediately route to the applicable coverage entity.

The displays on any Plexus Key Telephones will indicate **Do Not Disturb** and the LED on the **DND/FWD** button will appear solid red when DND mode is activated.

8.68.3 Operation

A. From a Plexus key telephone:

- To activate, press **DND/FWD**.
- To deactivate, press **DND/FWD**.

B. From an ordinary phone:

- To activate, enter **F11.**
- To deactivate, enter **F10**.

8.68.4 Administration

Users activate and deactivate DND mode from their station.

8.69 Do Not Disturb (DND) Override

8.69.1 Description

Do Not Disturb (DND) Override allows a user to override the DND mode of another user. Refer also to“Do Not Disturb (DND)” - section 8.68 .

8.69.2 Interactions

N.A.

8.69.3 Operation

A. From a Plexus key telephone:

- After receiving a DND error tone, press **OVERRIDE**.

or

- After receiving a DND error tone, press **FLASH** + **F12**.

B. From an ordinary phone:

- After receiving a DND error tone, **FLASH** + **F12**.

8.69.4 Configuration

The DND Override privilege grants a user the right to override the DND mode of all users. Limited DND override privileges may be granted by designating specific users on the user's DND override list instead of granting the universal privilege.

Table 8 - 55 Do Not Disturb (DND) Override

Method	Path	Parameter	Default
Software	User: Privileges User: Override, DND-Override	DND Override Name	Not Granted None
Key Telephone	User: User Privileges	Can DND Override	No

8.69.5 Administration

N.A.

8.70 Duration Timer

Plexus Key Telephones (display model only) feature a duration timer which displays the duration of each external call on the system. The timer displays in the upper-right corner of the LCD and appears automatically at the station where a call is connected.

8.70.1 Interactions

Call duration is output with call history reporting information (see “Station Message Detail Recording (SMDR)” - section 8.124).

8.70.2 Operation

N.A.

8.70.3 Configuration

The duration timer does not appear on the LCD until the Outbound-Dial Timeout expires.

Table 8 - 56 Duration Timer

Method	Path	Parameter	Default
Software	Trunk: Features	Outbound-Dial Timeout	20 seconds
Key Telephone	Trunk: Trunk Number	Outbound Dial TO	20 seconds

8.70.4 Administration

N.A.

8.71 End-to-End Signaling: External

8.71.1 Description

Plexus systems can accept DTMF tones from either Plexus Key Telephones or ordinary phones and send them through the public telephone network (via the Central Office) for access to a remote computer, voice mail system, or other device. The system can even establish a call using pulse dialing and then pass through DTMF tones in order to perform inward call completion functions as necessary for telephone banking and external voice mail applications.

8.71.2 Interactions

N.A.

8.71.3 Operation

N.A.

8.71.4 Configuration

N.A.

8.71.5 Administration

N.A.

8.72 End-to-End Signaling: Internal

8.72.1 Description

Plexus systems are capable of sending and receiving DTMF tones between stations or other extension devices on the system such as external voice adapters. Applications of this feature include Plexus Integrated Voice Mail and third-party voice mail.

8.72.2 Interactions

N.A.

8.72.3 Operation

N.A.

8.72.4 Configuration

The duration of DTMF signals generated at the extension to connect to internal devices (e.g., third-party voice mail adjuncts) may be modified for each extension port.

Table 8 - 57 End-to-End Signaling: Internal

Method	Path	Parameter	Default
Software	Extension Port: Device	DTMF Generation ON-Time DTMF Generation OFF-Time	100 ms 100 ms
Key Telephone	N.A.		

8.72.5 Administration

N.A.

8.73 Fax Detection

8.73.1 Description

Plexus systems offer Fax Detection on all lines. Incoming faxes are automatically detected and sent to a fax machine or modem on the system. Refer also to “Call Progress Tones” - section 8.45.

8.73.2 Interactions

The system attempts to detect an incoming fax before performing any call routing.

8.73.3 Operation

N.A.

8.73.4 Configuration

Fax detection relies on installing a filter profile for CNG tone and assigning this filter to a tone detector. The filters and detectors are installed and activated by default. Once identified by the detector, faxes are routed according to routing instructions defined for each trunk on the system. Fax Detection is effectively disabled on a trunk for which there are no routing instructions.

Table 8 - 58 Fax Detection

Method	Path	Parameter	Default
Software	Trunk: Routing: Fax Call Progress: Filter Call Progress: Detector	Route Fax during Day Route Fax during Night Filter Profile Tone Type-CNG	None None Fax CNG Activated
Key Telephone	Trunk: Trunk Number	Route Fax Day Route Fax Night	0 0

Note
It is not necessary to have any form of auto-attendant present in order to use this feature. The system will automatically seize a trunk with an incoming fax call if that trunk is set to route fax calls.

Tip
Should a system fail to identify incoming faxes, try increasing the number of cycles for the Call Progress CNG detector.

8.73.5 Administration

N.A.

8.74 Flash: Central Office

8.74.1 Description

Plexus systems are capable of sending a *flash* signal to the Central Office. Flash signals may be issued from either a Plexus Key Telephone or an ordinary phone. Refer also to “Centrex compatibility” - section 8.59.

8.74.2 Interactions

N.A.

8.74.3 Operation

A. From a Plexus key telephone:

- Press **FLASH** + **F54**.

or

- Press a DSS key programmed as **F54.**

B. From an ordinary phone:

- **FLASH** + **F54**.

8.74.4 Configuration

The ability to send a *flash* signal to the CO is configured on a user-basis.

Contact the local telephone company to determine the required flash signal duration. To assure compatibility, adjust the CO Flash Period accordingly.

Table 8 - 59 Flash: Central Office

Method	Path	Parameter	Default
Software	User: Privileges Trunk Port: Features	CO Flash CO Flash Period	Not Granted 480 ms
Key Telephone	User: Privileges	Can Flash CO	No

8.74.5 Administration

N.A.

8.75 Flash - System

8.75.1 Description

From a Plexus Key Telephone, several Feature Codes require a preceding *flash* before implementation or require to be programmed on **DSS** keys.

From an ordinary phone, several system features such as Call Transfer, Call Conference, and Camp-On require a preceding *flash* signal as well as several Feature Codes.

Therefore, Plexus systems offer a system 'flash' signal on Key Telephones and a programmable 'flash' signal on ordinary phones.

8.75.2 Interactions

In order to use a **FLASH** button on an ordinary phone (i.e., instead of the flashhook), the system flash timing may need to be modified for compatibility. If a User has Off-Hook preference of a trunk or trunk group and the user's Dial Expiration Timeout is 0 seconds, Flash - System is disabled as well as all features that require a Flash on Ordinary Phones (see "Off-Hook Preference" - section 8.95).

8.75.3 Operation

A. From a Plexus Key Telephone:

- Press the **FLASH** key.

B. From an ordinary phone:

- Press the **FLASH** button

or

- Press and release the switch-hook

8.75.4 Configuration

The duration of a system *flash* signal is programmable for each ordinary phone extension.

Table 8 - 60 Flash: System

Method	Path	Parameter	Default
Software	Extension Port: SLT	Flash Period Enabled Flash Period	Yes 800 ms
Key Telephone	N.A.		

Note
If Flash Period is not enabled, Flash hooking on an Ordinary Phone will cause instant disconnection of the call.

8.75.5 Administration

N.A.

8.76 Flexible Numbering Plan

8.76.1 Description

Plexus systems offer a Flexible Numbering Plan which allows the System Administrator to define access digits for operator access, trunk access, port access, feature access, and DISA access according to the specific needs at a particular installation. The Flexible Numbering Plan also allows the System Administrator to define the range and width of numbers used for user, user group, trunk, trunk group, auto-attendant, and voice mail IDs.

8.76.2 Interactions

IVPs are restricted to a range between 2 and 4 digits, with a maximum value of 8999.

8.76.3 Operation

A. From a Plexus key telephone:

Table 8 - 61 Flexible Numbering Plan

To perform this action	Do this
Access the operator	Press **0** (the default).
Access a trunk (CO line)	Press **9** (the default).
Access a feature	Press * (the default) followed by the feature code.
Access a port	Press # (the default).
Connect a user, user group, trunk, trunk group, auto-attendant resource, or voice-mail resource	Enter the applicable ID.
Access DISA	Press * (the default)

8.76.4 Configuration

The dial-number range determines which digits can be used as User IDs, User Group IDs, Trunk IDs, Trunk Group IDs, Auto-Attendant IDs, and Voice IDs. IDs can consist of 2 to 4 digits as defined in the width parameter. For example, if the range is from 1 to 5 and the width is 3, 100-599 can be used for IDs.

The selected range must be contiguous in order to be valid. For example, if the access digit defaults were accepted, 1 to 9 would not be a valid range since 9 is reserved for trunk access. 1 to 8, on the other hand, would be a valid range as it is contiguous.

Table 8 - 62 Flexible Numbering Plan

0	Operator	6	Available
1	Available	7	Available
2	Available	8	Available
3	Available	9	Trunk
4	Available	*	Feature
5	Available	#	Available

Table 8 - 63 Flexible Numbering Plan

Method	Path	Parameter	Default
Software	System Parameters: Dial Plan	Access Digit for Operator Access Digit for Trunk Access Digit for Port Access Digit for Feature Access Digit for DISA Dial-Number Range From Dial-Number Range to Dial-Number Range Width	0 9 # * * 1 4 3
Key Telephone	N.A.		

Note
Users are connected to their assigned operator and trunk Pilot-ID when they press the corresponding access digit (see “Multiple Operators” - section 8.89 and “Trunk Allocation” - section 8.132).

8.76.5 Administration

Users can press an access digit or enter an ID to connect to the corresponding resource.

See “Call Distribution Management” - section 8.33.

8.77 Group Listening

8.77.1 Description

Group Listening is a special speakerphone mode available on Plexus Key Telephones. With group listening activated, the handset and the speakerphone speaker operate simultaneously. This allows a single user to take advantage of the clarity and duplex capabilities of the handset while enabling several others to listen to the conversation through the speaker. Group Listening is useful in cases where only one person needs to speak and several people need to listen.

8.77.2 Interactions

Group Listening is not available in Headset Mode.

8.77.3 Operation

A. From a Plexus key telephone:

Table 8 - 64 Group Listening

To perform this action	Do this
Activate Group Listening	Press and hold down the **SPKR** button for two seconds while talking in the handset mode. The **SPKR** button illuminates slightly before group listening is enabled.
Deactivate Group Listening	To enable a private handset conversation, press **SPKR**. To enable a full speakerphone conversation, place the handset on hook.

Note
Depending on speaker volume and the acoustics of your office, it may be advisable to turn the group listening feature off before hanging up. This will eliminate a momentary squeal.

8.77.4 Configuration

N.A.

8.77.5 Administration

Group Listening is activated and deactivated as needed by the Plexus Key Telephone user.

8.78 Group Monitor Activity

8.78.1 Description

To Monitor a user groups, a **DSS** key must be programmed to monitor the user group. This key will flash when calls are ringing to the group (fast red flash) or when a call is in group's Hold Queue (slow flash red). You will also be able to retrieve group statistics by pressing the SCREEN key followed by the **DSS** key mapped to that group.

If you have the appropriate privileges, other group monitoring features will be available:

- You may also receive a Call Notify alert if a call is ringing to the group or a call is in the Hold Queue (see “Call Notify” - section 8.41).
- You may also retrieve calls that are ringing to the user group or are in the user group's Hold Queue (see “Universal Retrieve” - section 8.135 or “Call Retrieve” - section 8.50).

8.78.2 Interactions

Any user can access information about user groups. Pressing **SCREEN** and the **DSS** key mapped to a user group displays: _Queued (number of people currently on hold in the Queue), _Min (the time in minutes the oldest call has been in the Queue), _Mbrs (the number of users logged on to the group), _+ (total calls answered), _- (Queued calls that disconnected), and _? (calls that pressed '#' to leave a message). If the user group does not have Hold Queue enabled, all Hold Queue information will remain at 0.

8.78.3 Operation

N.A.

8.78.4 Configuration

N.A.

8.78.5 Administration

N.A.

8.79 Hands-Free Call

8.79.1 Description

Hands-Free Call allows a Plexus Key Telephone user with the appropriate privileges to place hands-free calls to other Plexus Key Telephone users. A Hands Free Call causes the target user's key telephone to automatically go off-hook in speakerphone mode. Refer also to "Auto-Answer" - section 8.13 and "Hands-Free Block" - section 8.80.

8.79.2 Interactions

Hands-Free Call does not require the target user to have Auto-Answer activated. The hands-free state is initiated when by the user placing the call.

If the called party is busy or in DND, the hands-free call will fail and the calling user will hear busy or DND error tone. The calling user may override at this point if they have Intrude or DND Override privileges (refer to chapter 6, "Software Configuration").

8.79.3 Operation

A. From either a Plexus key telephone or an ordinary phone:

- Enter **F22** + User ID.

8.79.4 Configuration

Table 8 - 65 Hands-Free Call

Method	Path	Parameter	Default
Software	User: Privileges	Hands-Free Call	Enabled
Key Telephone	User: User Privileges	Can Hands Free	Yes

8.79.5 Administration

N.A.

8.80 Hands-Free Block

8.80.1 Description

Hands-Free Block allows a user to block Hands-Free Calls from other users. When activated all Hands-Free Calls ring as a normal call. Hands-Free Block is useful in cases where a user needs a certain level of privacy not afforded by a Hands-Free Call. Refer also to “Hands-Free Call” - section 8.79 .

8.80.2 Interactions

N.A.

8.80.3 Operation

A. From a Plexus key telephone:

Table 8 - 66 Hands-Free Block

To perform this action	Do this
Enable Hands-Free Block	Press **F37**.
Disable Hands-Free Block	Press **F36**.

8.80.4 Configuration

N.A.

8.80.5 Administration

N.A.

8.81 Headset Mode

8.81.1 Description

Plexus Key Telephones accommodate the use of a headset with a special operating mode called Headset Mode. Headset Mode is similar to speakerphone operation in that the SPKR button serves as the handset switchhook. However, in Headset Mode, the conversation is exchanged through the headset earpiece and microphone.

8.81.2 Interactions

N.A.

8.81.3 Operation

For one touch switching from Headset to Handset Mode. DSS Key 17 must be programmed as **Utility: Use Headset**. The user will press this button to switch between modes. A solid red light indicates Headset Mode. No light indicates Handset Mode. Users can switch modes in the middle of a call.

Headset Mode makes use of the **SPKR** button instead of the handset switchhook for going off-hook and hanging up.

To mute a call while in Headset Mode, the user must utilize the mute button on the headset base as the Mute feature on the key telephone is only compatible with non-amplified headsets that use an electronic microphone.

8.81.4 Configuration

A. From a Plexus key telephone:

- Program DSS Key 17 as **Utility: Use Headset**. The button will be lit red for Headset Mode or not lit for Handset Mode.

Note
This feature is only available for Version 2.0 or later Plexus key telephones.

or

Table 8 - 67 Headset Mode

To perform this action	Do this
Enable Headset mode	Press the **Override** key.
Change Use Headset to Yes	Press the **XFER** key.

Note
This feature is only available for Version 1.01 or later Plexus key telephones.

8.81.5 Administration

N.A.

8.82 Home Station

8.82.1 Description

The Home Station feature allows the System Administrator to assign a user to a particular extension port. This assignment establishes a default or home station where the user's calls are directed in cases where the user is neither logged in nor signed in.

8.82.2 Interactions

If a user activates Call Forwarding: Immediate (All) before logging out, calls will follow the Call Forward instructions instead of routing to the home station.

Refer also to "Call Forwarding" - section 8.35.

8.82.3 Operation

N.A.

8.82.4 Configuration

Table 8 - 68 Home Station

Method	Path	Parameter	Default
Software	Extension Port: General	Port Assignment	None
Key Telephone	Extension Ports: Port Number	Bound User ID (Port Assignment)	0

8.82.5 Administration

N.A.

8.83 Hospitality Package

8.83.1 Description

Plexus systems include several features specifically designed to address the needs of hotels and motels. These features are discussed in chapter 18, "Hotel / Motel Package".

8.84 Intercept Service

8.84.1 Description

Also Known as "Reassurance Messages". See also “Busy Queuing - User” - section 8.21 and “Busy Queuing - User Group” - section 8.22.

8.84.1.1 User

Busy Queuing for a user or user group can be set up such that the queue is intercepted periodically with a default system message, a custom message, or a connection to another user on the system. The default system message provides a point-in-queue indication so that the caller knows how many calls are ahead of them. While custom messages cannot take advantage of the point-in-queue indication, they should provide reassurance to callers, letting them know that the system is aware that they are still holding. If the intercept is directed to another user on the system, that user can update the caller on the availability of the targeted user. Busy Queuing occurs when a call is Camped-On or Parked for the busy user; it is not available to calls placed on hold.

Note
Custom messages are recorded as a greeting in the case of an AAI or as a Play Message node in the case of an IVP. Refer to chapter 9, "Integrated Voice Processor" or chapter 15, "Automated Attendant Interface" for more information.

8.84.1.2 User Group

Intercept service is also available for user groups which have queuing enabled.

8.84.2 Operation

N.A.

8.84.3 Configuration

8.84.3.1 User

To enable intercept service for a user, enable and define an intercept period and assign an intercept pilot.

8.84.3.2 User Group

To set up intercept service for a user group, enable and assign an intercept pilot.

Table 8 - 69 Intercept Service

Method	Path	Parameter	Default
Software	User: Assignment, Support User Group: Distribution	Intercept Period Intercept Pilot Intercept Period Intercept Pilot	30 seconds None 30 seconds None
Key Telephone	User: User Attributes User Group: User Group Attributes	Intercept Period Intercept Pilot ID Intercept Period Intercept Pilot ID	30 seconds 0 30 seconds 0

8.84.4 Administration

N.A.

8.85 Key Telephone Configuration

8.85.1 Description

Many programmable parameters on Plexus systems can be configured using a Plexus Key Telephone (display model only). Upon entering the correct password and activating programming mode, several menus and parameters are displayed on the key telephone's LCD. Parameters are defined by selecting preprogrammed options or by entering alphanumeric information through the various buttons and keys.

See chapter 7, "Administration by Keyphone" for more information.

8.85.2 Interactions

N.A.

8.85.3 Operation

N.A.

8.85.4 Configuration

In order to activate programming mode and configure the system using a Plexus Key Telephone, a user must correctly enter the system-defined Admin password.

Table 8 - 70 Key Telephone Configuration

Method	Path	Parameter	Default
Software	System Parameters: General	Admin Password	123456
Key Telephone	System Parameters	Admin Password	123456

8.85.5 Administration

N.A.

8.86 Least Cost Routing

8.86.1 Description

Least Cost Routing is used to direct outgoing calls to specific trunks and/or to pre-dial specific PIC codes to utilize the most cost-efficient route for each call

8.86.2 Interactions

When user dials '9' for a Trunk group, has user-Features-Options-Least Cost Routing Enabled, and then dials the number, the system accesses System Parameters-Route Map-Outbound: LCR Map to determine on which Trunk to Route the call. Under Dialed Number, enter the prefix for the desired dial string. Double click in the Dialed Number field and a new window will popup with 'LCR Definition for #[prefix]". In this window you have fort time periods (see System Parameters-Time-Band) that you can route calls: Morning, Afternoon, Evening, and Night. For each of these time frames (Time Bands) you can define a Trunk or Trunk group to route the call onto. If desired you can also define a PIC code for the system to dial on the Trunk before dialing the call. The PIC Codes are entered in System Parameters-Route Map-PIC Codes. Default: Blank

8.86.3 Operation

N.A.

8.86.4 Configuration

See chapter 14, "Least Cost Routing".

8.86.5 Administration

N.A.

8.87 Message Waiting Indication

8.87.1 Description

Plexus systems are capable of providing Message Waiting Indication to both Plexus Key Telephones and ordinary phones. Visual indication, in the form of a flashing red LED (on the MSG button), is provided on Plexus Key Telephones. Audible indication by means of a stutter dial tone is provided on ordinary phones. Message Waiting Indication is initiated manually by a user on the system such as a receptionist or automatically by the Integrated Voice Processor (IVP) on systems with the Plexus Integrated Voice Mail option.

Note
While the audible indication is always available on ordinary phones, a visual indication may be provided on ordinary phones that have a message waiting lamp. However, the lamp must be compatible with the Plexus signaling format.

8.87.2 Interactions

Ordinary phone users may check their messages by issuing the appropriate feature code (**F52**) at system dial tone. Plexus Key Telephone users check their messages by pressing the **MSG** button.

8.87.3 Operation

From either a Plexus key telephone or an ordinary phone:

Table 8 - 71 Message Waiting Indication

To perform this action	Do this
Manually activate a user's Message Waiting Indication	Enter **F51** + User ID.
Manually deactivate a user's Message Waiting Indication	Enter **F50** + User ID.

8.87.4 Configuration

Pressing the **MSG** button on a Plexus Key Telephone or issuing the feature code (**F52**) for checking messages on an ordinary phone, connects the user to their Pilot-ID for Voice-Messages. On systems with the Plexus Integrated Voice Mail option, the Pilot-ID is a Voice ID created for checking messages. On systems without Plexus Integrated Voice Mail, the Pilot-ID will typically be a receptionist who manually takes messages for other users or an answering service.

Table 8 - 72 Message Waiting Indication

Method	Path	Parameter	Default
Software	User: Assignment: Message Access	Pilot-ID for Message Access	None
Key Telephone	User: User Attributes	Message Pilot ID	0

8.87.5 Administration

N.A.

8.88 Multiple Auto-Attendants

8.88.1 Description

Plexus systems can support multiple auto-attendants enabling support for several different departments or tenants and various other special applications. A multiple auto-attendant configuration can be achieved through the use of several Auto-Attendant Interface (AAI) cards or through the use of the ten available Route Points on an Integrated Voice Processor (IVP).

Refer to chapter 9, "Integrated Voice Processor" or chapter 15, "Automated Attendant Interface" for more information.

8.88.2 Interactions

N.A.

8.88.3 Operation

N.A.

8.88.4 Configuration

N.A.

8.88.5 Administration

N.A.

8.89 Multiple Operators

8.89.1 Description

Plexus systems are flexible with regard to operator definition and assignment. Each user may be assigned a different operator thus supporting multiple operator applications to serve different departments within an organization.

Refer also to “Operator: Personal” - section 8.98 and “Operator: System” - section 8.99.

8.89.2 Interactions

N.A.

8.89.3 Operation

N.A.

8.89.4 Configuration

N.A.

8.89.5 Administration

N.A.

8.90 Multiple Tenant Service

8.90.1 Description

Plexus systems are capable of supporting multiple tenant applications through use of the Flexible Numbering Plan, multiple operators, trunk-specific call routing, multiple auto-attendants, and several programmable parameters.

8.90.2 Interactions

N.A.

8.90.3 Operation

N.A.

8.90.4 Configuration

N.A.

8.90.5 Administration

N.A.

8.91 Music-On-Hold

8.91.1 Description

The Music-On-Hold feature provides background music to held calls. External music sources are connected to a standard RCA port (labeled **MOH**) on the Digital Switch Processor (DXP) card (Plexus) or the motherboard (Micro-Plexus).

Tip
Music-On-Hold may be used to play a continuous recording featuring product and/or company information.

8.91.2 Interactions

In the case of a linked system, the music source must be connected to the Digital Switch Processor (DXP) cards on both cabinets.

Use of either the Music-On-Hold or the paging interface (or both) on a Micro system, utilizes the resources of extension port #8 on the motherboard and makes this port unavailable for other devices.

8.91.3 Configuration

8.91.3.1 Plexus

No configuration is necessary to utilize the Music-On-Hold feature. If an external music source is properly connected to the DXP card, music is supplied to held calls.

8.91.3.2 Micro-Plexus

In order to utilize Music-On-Hold on a Micro-Plexus system, extension port #8 must be disabled. Note that this particular port (Unit ID=0, Slot ID=0, Port ID=7) is disabled by default.

Table 8 - 73 Music-On-Hold

Method	Path	Parameter	Default
Software	Extension Port: General	Port Enabled	Disabled
Key Telephone	N.A.		

Note
Virtually any audio device (e.g., radio, cassette player) may be connected to the system.

8.91.4 Administration

N.A.

8.92 Mute

8.92.1 Description

Plexus Key Telephones offer a Mute feature which disables the microphone (in both handset and speakerphone mode) when activated. Mute prevents the user from being heard by other parties to the call, but does not prevent the user from hearing the other parties.

8.92.2 Interactions

Mute is automatically deactivated upon termination of the call.

Mute is unavailable in Headset Mode.

8.92.3 Operation

A. From a Plexus key telephone:

- To activate, press **MUTE**.
- To deactivate, press **MUTE** again.

8.92.4 Configuration

N.A.

8.92.5 Administration

N.A.

8.93 Name Labeling (System Resources)

8.93.1 Description

All of the various system resources or entities (e.g., users, user groups, trunks, trunk groups) can be given an informal alphanumeric name label. These names are useful in identifying the particular resource throughout the configuration process. Assigned user names can be displayed on a Plexus Key Telephone user's display (if available) and are useful in identifying the calling user on an internal call (see "Calling Party Display: Internal" - section 8.56).

8.93.2 Interactions

N.A.

8.93.3 Operation

N.A.

8.93.4 Configuration

Table 8 - 74 Name Labeling

Method	Path	Parameter	Default
Software	User: General User Group: General Trunk: General Trunk Group: General Auto-Attendant: General Voice Processor: General	Name Name Name Name Name Name	User # User Group # Trunk # Trunk Group # _____ _____
Key Telephone	User User Group Trunk Trunk Group Auto Attendant Int Voice Mail	Name Name Name Name Name Name	

8.93.5 Administration

N.A.

8.94 Non-Volatile Configuration Memory

8.94.1 Description

Plexus systems utilize non-volatile or nv RAM to store system programming. This means that in the event of a power outage, the unit will retain its programming and not require reconfiguration. Even over extended periods with no power (on the order of 10 years), the configuration will remain intact.

This feature does not mean that the system will function during the power outage, rather that when power returns the unit will be ready for operation.

See “Battery Backup / UPS” - section 8.18 for details on operating the system during a power failure.

8.94.2 Interactions

N.A.

8.94.3 Operation

N.A.

8.94.4 Configuration

N.A.

8.94.5 Administration

N.A.

8.95 Off-Hook Preference

8.95.1 Description

The Off-Hook Preference feature enables user parameters to be configured such that each time the user goes off-hook at the station to which they are logged in, they are connected to another user, user group, trunk, trunk group, auto-attendant, voice mail system, or paging facility. The system can even be configured to connect to a trunk and dial an external telephone number, thereby immediately connecting to a remote telephone.

The Plexus system allows for 'warm' Off Hook Preference. When Off Hook Preference is assigned, user's Dial Expiration defaults to '0' seconds. To enable 'warm' Off Hook Preference, the user's Dial Expiration must be reset to the desired number of seconds before Off Hook Preference begins. If the user makes a call during this time, Off Hook Preference will be disabled until the call is complete. For 'hot' Off Hook Preference, leave the Dial Expiration at '0' seconds.

Note
The Off-Hook Preference feature is typically used with Plexus Door Phones.

8.95.2 Interactions

An ordinary telephone with 'hot' Off-Hook Preference will be unable to do anything else. This includes accessing all features that require a flash hook or internal dialing, making internal calls, paging, etc.

8.95.3 Operation

N.A.

8.95.4 Configuration

The Off-Hook Preference entity (user, trunk, or page) must first be selected. If user is selected, a Pilot-ID assignment must be made from among the available users and user groups. If trunk is selected, a Pilot-ID assignment must be made from among the available trunks and trunk groups and optionally the external dial number option may be enabled and defined with a remote telephone number. Finally, if page is selected, the paging facility must be selected (i.e., internal, external, or answer) and if the internal facility is selected a Pilot-ID assignment must be made (e.g., a user or user group).

Table 8 - 75 Off-Hook Preference

Method	Path	Parameter	Default
Software	User: Assignment: Off-Hook Preference User: Features: Timeouts	None, User, Trunk, Page User, Pilot-ID Trunk, Pilot-ID Trunk, Use External Dial Number Trunk, Telephone Number Page, Pilot-ID Dial Expiration Page Preference	None None None Disabled _____ 20* None Internal
Key Telephone	User: User Attributes	Off Hook Preference Off Hook Pilot ID Off Hook Dial #	None 0 __

8.95.5 Administration

N.A.

8.96 On-Hook Dialing

8.96.1 Description

On-Hook Dialing (or Hot Dialing) allows a Plexus Key Telephone user to place calls without lifting the handset. To use On-Hook Dialing, the user simply begins dialing without lifting the handset. The key telephone will automatically go off-hook in speakerphone mode and begin dialing the entered digits.

8.96.2 Interactions

N.A.

8.96.3 Operation

A. From a Plexus key telephone:

Table 8 - 76 On-Hook Dialing

To perform this action	Do this
Dial	Begin pressing digits on the keypad.
Disconnect	Press the illuminated **SPKR** button.

8.96.4 Configuration

N.A.

8.96.5 Administration

N.A.

8.97 Operator Console support

8.97.1 Description

Plexus systems support the Plexus DSS-60/DDSS60 Operator Console for extended Direct Station Select (DSS) and Direct Trunk Select (DTS) applications such as a receptionist or operator station. The operator console is also useful for front-desk use at a hotel or motel (see “Hospitality Package” - section 8.83).

8.97.2 Interactions

N.A.

8.97.3 Configuration

Operator consoles are programmed through a host Plexus Key Telephone. Each key telephone can support up to seven consoles. The host key telephone and all of the supported consoles must be wired to the same Analog Extension Interface (AEI) or Digital Extension Interface (DEI) peripheral card.

8.97.4 Administration

N.A.

8.98 Operator: Personal

8.98.1 Description

An operator assignment is made for each user. The defined operator is the entity that the user will be connected to when they press the operator access digit (see “Flexible Numbering Plan” - section 8.76). The operator assignment may be an individual colleague, department receptionist, or operator group.

8.98.2 Interactions

On systems with the Plexus Integrated Voice Mail option, callers can access a user’s operator by pressing **9** upon entering the user’s voice mailbox (i.e., during the user’s greeting). From the system dial tone on the Plexus, the user may access the operator by pressing **0**.

8.98.3 Operation

N.A.

8.98.4 Configuration

Table 8 - 77 Operator: Personal

Method	Path	Parameter	Default
Software	User: Assignment: Group Access	Pilot-ID for Operator	None
Key Telephone	User: User Attributes	Operator Pilot ID	0

8.98.5 Administration

N.A.

8.99 Operator: System

8.99.1 Description

A default operator assignment is made at the system level (i.e., System Parameters). The defined system operator is the entity that an unassigned station is connected to when the operator access digit (see “Flexible Numbering Plan” - section 8.76) is pressed. The system operator also serves as the default target for calls received on trunks with no defined routing. The system operator assignment may be an individual colleague, department receptionist, or operator group. Refer also to “Multiple Operators” - section 8.89.

8.99.2 Interactions

N.A.

8.99.3 Operation

N.A.

8.99.4 Configuration

Table 8 - 78 Operator: System

Method	Path	Parameter	Default
Software	System Parameters: Feature, Assignment	Default Assignment Operator	None
Key Telephone	N.A.		

8.99.5 Administration

N.A.

8.100 PBX Operation

8.100.1 Description

Plexus systems are hybrid systems in that they can be configured to behave like either a traditional key system or a Private Branch Exchange (PBX) system. Users can access outside lines by pressing a Direct Trunk Select (**DTS**) key or by pressing **9**. Additionally, the system can be configured such that Plexus Key Telephone users with the appropriate privileges receive visual indication of ringing stations. The system can even be configured to simultaneous ring every member of a user group. Finally, Plexus systems support both key telephones and ordinary analog telephones and telephone devices (e.g., modems, fax machines, credit card scanners, etc.).

Note
Plexus systems are fully-protected hybrid systems and are therefore categorized as MF equipment according to FCC registration. Installers must provide this information to the telephone company at the time of installation. Note that MF equipment may incur a higher monthly tariff than KF or Key Telephone equipment.

8.100.2 Interactions

N.A.

8.100.3 Operation

N.A.

8.100.4 Configuration

The Ring On and Off and *flash* signal periods can be defined for each extension port to which an ordinary analog telephone will be connected.

Table 8 - 79 PBX Operation

Method	Path	Parameter	Default
Software	Extension Port: SLT	Ring ON Ring OFF Flash Period	1 second 3 seconds 800 ms
Key Telephone	N.A.		

8.100.5 Administration

N.A.

8.101 Page: All Points

8.101.1 Description

The All Points Paging facility allows users to page through both the Plexus Key Telephones and a customer-provided external (e.g., overhead) paging system. An All Points Page cannot be limited to the key telephones of a specific user group.

8.101.2 Interactions

All of the interactions that apply to Internal and External Pages, also apply to an All Points Page.

8.101.3 Operation

From either a Plexus key telephone or an ordinary phone:

- To place an all points page, enter **F26**.

8.101.4 Configuration

Users must be granted the Page privilege in order to make use of any paging facility (internal, external, all points). The Page Duration Limit controls the length of a page announcement.

Table 8 - 80 Page: All Points

Method	Path	Parameter	Default
Software	User: Privileges System Parameters: Feature, Timeouts	Page Page Duration Limit	Enabled 30 seconds
Key Telephone	User: User Privileges	Can Page	Yes

8.101.5 Administration

N.A.

8.102 Page: Answer

8.102.1 Description

Answer Page allows a user to respond to a page by being automatically connected to the paging party. An Answer Page may be performed while the page is in progress or immediately following the page.

8.102.2 Interactions

Each time a user places a page, the Answer Page facility is reset to apply to the user who placed the most current page.

If the paging party is unavailable when an Answer Page is performed, the automatic connection will fail.

8.102.3 Operation

From either a Plexus key telephone or an ordinary phone:

- To answer a page, enter **F27**.

8.102.4 Configuration

N.A.

8.102.5 Administration

N.A.

8.103 Page: Block

8.103.1 Description

Page Block allows a user to block the broadcast of pages through their Plexus Key Telephone. Page Block is useful in cases where a user does not want to be disturbed by pages.

8.103.2 Interactions

N.A.

8.103.3 Operation

A. From a Plexus key telephone:

Table 8 - 81 Page: Block

To perform this action	Do this
Block paging	Enter **F29**.
Unblock paging	Enter **F28**.

8.103.4 Configuration

N.A.

8.103.5 Administration

N.A.

8.104 Page: External

8.104.1 Description

The External Paging facility allows users to page through a customer-provided external (e.g., overhead) paging system. The external paging system is connected to the RCA port, labeled **Pager Output**, on the Digital Switch Processor (DXP) card.

Warning
Some external paging systems can give voltage feedback to the Plexus, possibly damaging the Plexus system. See chapter 4, "Installation" for information on proper external pager system installation.

8.104.2 Interactions

In the case of a linked system, the external paging system must be connected to the Digital Switch Processor (DXP) cards on both cabinets.

Use of either the paging or the Music-On-Hold interface (or both) on a Micro system, utilizes the resources of extension port #8 on the motherboard and makes this port unavailable for other devices.

While a page is in progress or immediately following a page, a user can enter the Page Answer feature code to be connected to the paging party.

8.104.3 Operation

A. From either a Plexus key telephone or an ordinary phone:

Table 8 - 82 Page: External

To perform this action	Do this
Place a page over an external facility	Enter **F25**.
Place a page to all points (keyphones and external)	Enter **F26**.

8.104.4 Configuration

Users must be granted the Page privilege in order to make use of any paging facility (internal, external, all points). The Page Duration Limit controls the length of a page announcement.

Table 8 - 83 Page: External

Method	Path	Parameter	Default
Software	User: Privileges System Parameters: Feature, Timeouts	Page Page Duration Limit	Enabled 30 seconds
Key Telephone	User: User Privileges	Can Page	Yes

8.104.5 Administration

N.A.

8.105 Page: Internal

8.105.1 Description

The Internal Paging facility allows users to page through Plexus Key Telephone speakers. An Internal Page may be issued to all key telephones or to the key telephones of a specific user group only.

Paging to a specific user group only is also known as "Zone Paging"

8.105.2 Interactions

While a page is in progress or immediately following a page, a user can enter the Page Answer feature code to be connected to the paging party.

8.105.3 Operation

A. From either a Plexus key telephone or an ordinary phone:

- To page over *all* key telephones, enter **F23**.
- To page over a specific group of key telephones, enter **F24** + User Group ID.

8.105.4 Configuration

Users must be granted the Page privilege in order to make use of any paging facility (internal, external, all points). The Page Duration Limit controls the length of a page announcement.

Table 8 - 84 Page: Internal

Method	Path	Parameter	Default
Software	User: Privileges System Parameters: Feature, Timeouts	Page Page Duration Limit	Enabled 30 seconds
Key Telephone	User: User Privileges	Can Page	Yes

8.105.5 Administration

N.A.

8.106 Pager Notification

8.106.1 Description

The Pager Notification feature provided with the Plexus Integrated Voice Mail option enables user's to receive notification on a digital pager each time they receive a new voice mail message in their mailbox.

8.106.2 Interactions

The system utilizes a line according to the user's assigned trunk Pilot-ID to place the call to the pager (see "Trunk Allocation" - section 8.132) unless the notification is for a mailbox that is not associated with a user (e.g., a user group mailbox) in which case the system utilizes the system default trunk assignment.

8.106.3 Operations

N.A.

8.106.4 Configuration

The Integrated Voice Processor (IVP) must be configured correctly for this feature to work. Refer to the optional IVP section in the product manual for more information.

8.106.5 Administration

N.A.

8.107 Password Security

8.107.1 Description

Plexus systems utilize password security in the following areas: Authorization Management, User Alias, DISA/RSS, Key Telephone Configuration, and System Administrator Privilege. The feature and applicable password are as follows:

Table 8 - 85 Password Security

Feature	Applicable password
Authorization Management	User/User Group
User Alias	User
DISA/RSS	DISA
Key Telephone Configuration	Admin
Software Link Access	Admin
Software Operation Access	Plexus Admin Program
System Administrator Privilege	Admin

8.107.2 Interactions

See Software Configuration section for further information.

8.107.2.1 Random Password Generation for users

At user-Features-Settings-Password, the program allows several choices when right clicking the mouse;

Replicate ALL: after typing in a password, the same password can be given to all users.

Append ALL- user-ID: Sets all users' passwords to their user number.

Apply ALL Random Setting: Allows all users to receive a random password. The random password can be set from 1 to 12 digits long.

Apply Random Setting: Allows the specific user to receive a random password that can be set from 1 to 12 digits long.

8.107.2.2 Random Password Generation for user groups

The system allows for a password to be given to a user group. This setting is for users logging into the group (see “Agent Log Off / Agent Log On” - section 8.5).

Apply ALL Random Setting: Allows all user groups to receive a random password. The random password can be set from 1 to 12 digits long.

Apply Random Setting: Allows the user group to receive a random password that can be set from 1 to 12 digits long.

8.107.2.3 Password for DISA/RSS

The DISA password must be entered in a DSS enabled trunk to access DISA/ RSS privileges. This prevents unauthorized use of system resources and telephone lines.

8.107.2.4 Password for Key Telephone Configuration

The System Administrator Password must be entered to enter the system programming mode of Key Telephones.

8.107.2.5 Password for Software Link Access

The System Administrator Password must be entered to properly link with the system. The wrong password will cause an access denied message. This is to prevent unauthorized changes to the system.

8.107.2.6 Software Operation Access

A password can be set for running Plexus Administrator software. The password is modified through File-Set Password. This is to prevent unauthorized changes to the saved configuration files.

8.107.2.7 System Administrator Privilege

The System Administrator Privilege gives users access rights to access Features or access Programming modes.

Note
Users with administrate privilege will automatically have the privilege enabled on Log In.

8.107.3 Operation

N.A.

8.107.4 Configuration

Passwords are assigned during the initial configuration of the system.

Table 8 - 86 Password Security

Method	Path	Parameter	Default
Software	User: Features, Settings User Group: General System Parameters: General	Password Password Admin Password DISA Password	None None 123456 123456
Key Telephone	User: User Attributes	Password	None

8.107.5 Administration

Passwords may contain up to 12 digits.

8.108 Port Management: Line and Station

8.108.1 Description

Extension and trunk ports can be enabled or disabled. To utilize an enabled port, it must be properly configured through either software or key telephone configuration. A user or trunk is also typically assigned to each enabled extension and trunk port, respectively.

8.108.2 Interactions

Trunk ports without assigned trunks are automatically routed to the defined system operator (see "Operator: System" - section 8.99). Ports may be enabled or disabled only when logged on as System Administrator.

8.108.3 Operation

8.108.3.1 Activating Ports

A. From either a Plexus key telephone or an ordinary phone:

- Log in as System Administrator by entering **F91** and the password.
- Enter **F97** + port ID.

8.108.3.2 Deactivating Ports

A. From either a Plexus key telephone or an ordinary phone:

- Log in as System Administrator by entering **F91** and the password.
- Enter **F98** + port ID.

8.108.4 Configuration

Table 8 - 87 Port Management: Line and Station

Method	Path	Parameter	Default
Software	Extension Port: General Trunk Port: General	Port Enabled Port Assignment Port Enabled Port Assignment	Enabled None Enabled None
Key Telephone	Extension Port: Attributes	Bound User ID (Port Assignment)	None

Note
In the case of a linked system, key telephone configuration of trunk and extension ports must be performed from an extension on the same cabinet as the trunk or extension port being configured.

8.108.5 Administration

N.A.

8.109 Privacy Management

8.109.1 Description

Privacy Management deals with a user's ability to intrude on or monitor other users that are either busy or in DND mode. There are several features that deal with user privacy. They are Do Not Disturb (DND), Do Not Disturb (DND) Override, Call Intrude, Call Monitor, and Silent Monitoring. Intrude, monitor, and override privileges are granted by the System Administrator. Privileges may be granted on a limited basis or on a system-wide basis. Users can even grant intrude and monitor privileges to other users on a per conversation basis (see "Privacy Release" - section 8.110).

8.109.2 Interactions

N.A.

8.109.3 Operation

N.A.

8.109.4 Configuration

N.A.

8.109.5 Administration

N.A.

8.110 Privacy Release

8.110.1 Description

Privacy Release enables a user to grant intrude and monitor privileges to any user on the system on a per conversation basis. This feature is useful in situations where a user would like one or more of their colleagues to join or monitor their call in progress.

Note
When a user intrudes on a call, they become conferenced with the other parties on the call (see “Call Intrude” - section 8.39).

8.110.2 Interactions

N.A.

8.110.3 Operation

A. From a Plexus key telephone:

Table 8 - 88 Privacy Release

To perform this action	Do this
Allow calls to intrude using Privacy Release	During a conversation, press the **Override** key on the key telephone. Any incoming calls will receive a busy signal, and the **Override** key on a key telephone will start to flash. If the **Override** key is pressed, it will Intrude.
Allow calls to be monitored using Privacy Release	During a conversation, press the **Override** key on the key telephone. Any incoming calls will receive a busy signal, and their override key will start to flash. To monitor, the incoming caller must press **FLASH** + **F13** after receiving the busy signal.

B. From an ordinary phone:

Table 8 - 89 Privacy Release

To perform this action	Do this
Allow calls to intrude using Privacy Release	1. Press the **FLASH** button. 2. Enter **F12**.
Allow calls to be monitored using Privacy Release	1. Press the **FLASH** button. 2. Enter **F13.**

8.110.4 Configuration

N.A.

8.110.5 Administration

N.A.

8.111 Programmable keys: Key Telephone

8.111.1 Description

Plexus Key Telephones offer programmable keys which may be defined by the user or the System Administrator to perform any of several functions including Direct Station Select (DSS)/BLF, Direct Trunk Select (DTS)/BLF, automated feature and function code entry, and speed dial.

8.111.2 Interactions

N.A.

8.111.3 Operation

A. From a Plexus key telephone:

- Hold down the **OVERRIDE** key until **Use Headset** is displayed.
- Press **VOL ▲** until **Define Keys** is displayed.
- Press **SPKR** (not needed on 2.0 Key Telephones and above). Keys 01 to 20 will now be red.
- Press the desired **DSS** key. It will start to blink.
- Use the **XFER** key to scroll through the options to the desired setting for that key.
- Press **OVERRIDE** again to save the change.

8.111.4 Configuration

N.A.

Table 8 - 90 Programmable Keys: Key Telephone

Method	Path	Parameter	Default
Software	User: Features, Settings	Define Key… on Key-Telephone	N/A

Note
Software configuration of the programmable keys requires version 1.01 or more recent Plexus Key Telephone firmware.

Note
Instant update of a single Plexus Key Telephone through the software feature or verification of a single Plexus Key Telephone' programming requires version 2.0 or more recent Plexus Key Telephone firmware.

The firmware version is displayed on a display model key telephone when the telephone initializes. To initialize a key telephone, disconnect the base cord and then plug it back in, or use the feature code **F99**.

8.111.5 Administration

Programmable keys can be defined by entering programming mode on the key telephone or through software configuration. Typically, the initial configuration is performed through the software and subsequent modifications are made through programming mode on the key telephone (see the **PVT-30D/PDT-30D Telephone User's Guide** for more information).

8.112 Programmable Trunk Timing Parameters

8.112.1 Description

Plexus systems offer several programmable timing parameters providing significant flexibility. Many of the timing parameters, such as the Timeout on Exclusive Hold and the DISA Session Timeout, deal with user preferences and internal applications. These timing parameters are discussed with the related feature. Several other timing parameters deal with the interaction between the system and the public telephone network. These parameters may be modified to optimize compatibility with the local Central Office. Refer also to “CO Disconnect Supervision” - section 8.23 and “CO Pulse/Tone Dialing” - section 8.24.

8.112.2 Interactions

N.A.

8.112.3 Configuration

The Guard Duration is the period of time that a trunk is unavailable following the completion of a call.

The Ring Qualifier (RQ) is the duration necessary for voltage on the line to be considered a ring tone by the system. Voltage of a shorter duration is considered line noise.

Ring Silence (RS) is the duration of the interval between ring tones used by the system to determine that a call was terminated.

Seize Settling (SS) is the amount of time the system must wait once a trunk port has been opened (e.g., seizing a trunk) to reach the nominal operating condition. During this period, disconnect signals are ignored, and digit buffering is in effect.

Central Office Setup (OS) is the amount of additional time following Size Settling (SS) that the system must wait to allow the Central Office to reach the nominal operating condition. During this period, disconnect signals are ignored and digit buffering is in effect. Upon expiration of the OS timeout, the buffer is emptied and digits are sent to the Central Office.

Table 8 - 91 Programmable Trunk Timing Parameters

Method	Path	Parameter	Default
Software	Trunk Port: Features	Guard Duration RQ (Ring Qualifier) RS (Ring Silence) SS (Seize Settling) OS (Central Office Setup)	2,000 ms 200 ms 7,000 ms 1,000 ms 500 ms
Key Telephone	N.A.		

8.112.4 Administration

N.A.

8.113 Redial: Last Number

8.113.1 Description

Last Number Redial enables a user to instruct the system to redial up to 20 digits of the last number the user dialed.

Note
The feature only applies to external calls.

Note
Last Number Redial dials the digits a single time each time the user issues the feature code or presses the **REDIAL** button on a Plexus Key Telephone.

8.113.2 Interactions

The system utilizes a line according to the user's assigned trunk Pilot-ID (see "Trunk Allocation" - section 8.132) to redial the number. If no trunk assignment is made, the redial attempt will fail.

8.113.3 Operation

A. From a Plexus key telephone:

- Press **REDIAL**.

or

- Enter **F56**.

B. From an ordinary phone:

- Enter **F56**.

8.113.4 Configuration

N.A.

8.113.5 Administration

N.A.

8.114 Remote Call Forwarding

8.114.1 Description

The Remote Call Forwarding feature provided with the Plexus Integrated Voice Mail option enables user's to forward their calls, routed by the auto-attendant, to a remote telephone. The calls will be forwarded once the system tries to route the calls to the user's voice mailbox. If the number entered in Remote Call Forwarding does not answer or refuses the call, the system will prompt the caller to leave a message in the user's voice mailbox.

8.114.2 Interactions

The system utilizes a line according to the user's assigned trunk Pilot-ID to forward the call (see "Trunk Allocation" - section 8.132) unless the call is being forwarded for a mailbox that is not associated with a user (e.g., a user group mailbox) in which case the system utilizes the system default trunk assignment.

8.114.3 Operation

N.A.

8.114.4 Configuration

The Integrated Voice Processor (IVP) must be configured correctly for this feature to work. Refer to the optional IVP section in the product manual for more information.

8.114.5 Administration

Users activate and deactivate Remote Call Forwarding through their voice mailbox settings.

8.115 Remote Telephone Notification

8.115.1 Description

The Remote Telephone Notification feature provided with the Plexus Integrated Voice Mail option enables user's to receive notification at a remote telephone each time they receive a new voice mail message in their mailbox.

8.115.2 Interactions

The system utilizes a line according to the user's assigned trunk Pilot-ID to place the call to the remote telephone (see "Trunk Allocation" - section 8.132) unless the notification is for a mailbox that is not associated with a user (e.g., a user group mailbox) in which case the system utilizes the system default trunk assignment.

8.115.3 Operation

N.A.

8.115.4 Configuration

The Integrated Voice Processor (IVP) must be configured correctly for this feature to work. Refer to the optional IVP section in the product manual for more information.

8.115.5 Administration

Users activate and deactivate Remote Telephone Notification through their voice mailbox settings.

8.116 Remote Station Service (RSS)

8.116.1 Description

Remote Station Service (RSS) is an enhancement to DISA. RSS allows a user to log-in from a remote location using DISA. Once logged in to the system, the user will have access to their defined class of service as if they were logged in on-site, at an ordinary phone. Refer also to “Direct Inward System Access (DISA)” - section 8.63.

8.116.2 Interactions

To utilize RSS, the user must properly access DISA.

An RSS session is subject to the DISA Session Timeout.

8.116.3 Operation

A. From an external telephone:

- Call a DISA enabled line.
- Enter * + DISA password.
- Enter **F31** + user ID + user password.

Note
Users press ** to go on-hook and ## to send a *flash* signal to the system while using RSS.

8.116.4 Configuration

N.A.

8.116.5 Administration

N.A.

8.117 Remote System Management (RSM)

8.117.1 Description

The Remote System Management (RSM) feature enables Plexus systems to be remotely configured and otherwise managed using Plexus Administrator. In order to utilize RSM, modems must be connected to the applicable PC and to the remote system. The system modem is connected to the serial data port with a null modem cable. To initiate RSM, a call is placed from the PC's modem to the remote system modem. All functionality available with Plexus Administrator using a direct serial connection is also available via a modem connection. Refer to chapter 17, "Remote System Management" and chapter 6, "Software Configuration" for further details.

8.117.2 Interactions

N.A.

8.117.3 Operation

N.A.

8.117.4 Configuration

N.A.

8.117.5 Administration

N.A.

8.118 Serial Data Port

8.118.1 Description

Plexus systems provide a serial data port interface for use in uploading system configurations, Switch Operating System (XOS) upgrades, auto-attendant configurations, mailbox configurations, and Voice Processing System (VPS) upgrades and components (e.g., system prompt updates and additional languages). The serial port is located on the Digital Switch Processor (DXP) card on the Plexus and on the Micro-Plexus motherboard. Refer also to “About serial ports and COM ports” - section 6.27.1 and chapter 17, "Remote System Management.

8.118.2 Interactions

N.A.

8.118.3 Operation

N.A.

8.118.4 Configuration

N.A.

8.118.5 Administration

N.A.

8.119 Silent Monitoring

8.119.1 Description

Silent Monitoring allows a user, with the appropriate privileges, to monitor or eavesdrop on a busy line or user. The monitoring user will be able to listen to, but will not be heard by, the parties on the call. Additionally, with Silent Monitoring, no notification tone is issued to the parties on the call. The parties, therefore, are unaware that the call is being monitored. Silent Monitoring is available on internal, external, and conference calls. Refer also to "Call Monitor" - section 8.40.

Note
Check with local laws to ascertain the legality of Silent Monitoring.

8.119.2 Interactions

Calls on any type of hold (i.e., Consult, Exclusive, System) cannot be monitored.

When Plexus Key Telephone users attempt to connect to a busy line or user for which there are monitor rights, the LED on the **OVERRIDE** button will flash.

Plexus Key Telephone users can grant all other users the right to monitor, on a per conversation basis, by pressing **OVERRIDE** during a call. The associated LED will appear solid red and all other users (even those without monitor privileges) will be able to monitor the call.

8.119.3 Operation

A. From a Plexus key telephone :

- After receiving a busy signal, press **FLASH** + **F13**.
- or
- After receiving a busy signal, press a **DSS** key programmed with **F13.**

B. From an ordinary phone:

- After receiving a busy signal, **FLASH** + enter **F13**.

8.119.4 Configuration

The Monitor ALL privilege grants a user the right to monitor all users and trunks. Limited monitor privileges may be granted by designating specific users on the user's monitor list instead of granting the universal privilege.

The Silent Monitoring privilege grants a user the right to monitor without a notification tone. If this privilege is granted all monitoring performed by the user is in 'silent' mode.

Table 8 - 92 Silent Monitoring

Method	Path	Parameter	Default
Software	User: Privileges User: Override, Monitor	Monitor ALL Silent Monitoring Name	Not Granted Not Granted None
Key Telephone	User: User Privileges	Can Monitor Call	No

8.119.5 Administration

N.A.

8.120 Software (Windows-based) configuration

8.120.1 Description

All programmable parameters on Plexus systems can be configured using Plexus Administrator, the windows-based software configuration utility. Plexus Administrator presents all of the parameters in an organized and intuitive manner. The System Administrator enters data and selects options to develop a configuration file. Configuration files are uploaded to the system via a direct serial connection or through a modem connection. Plexus Administrator is also used to upload Switch Operating System (XOS) upgrades and Voice Processing System (VPS) upgrades and components.

Refer to chapter 6, "Software Configuration" for further details.

8.120.2 Interactions

N.A.

8.120.3 Operation

N.A.

8.120.4 Configuration

N.A.

8.120.5 Administration

N.A.

8.121 Speakerphone operation

8.121.1 Description

Plexus Key Telephones offer a built-in speaker and microphone designed for high-quality speakerphone operation. The speakerphone allows Plexus Key Telephone users to carry-on a telephone conversation without lifting the handset. The speakerphone also enables the valuable Group Listening feature. Refer also to “Group Listening” - section 8.77 and “On-Hook Dialing” - section 8.96.

8.121.2 Interactions

Speakerphone operation makes use of the SPKR button instead of the handset switchhook for going off-hook and hanging up.

When the handset is picked up, speakerphone operation is terminated.

Pressing **MUTE** during speakerphone operation prevents the user from being heard by the other parties in the call, but does not prevent the user from hearing the other parties.

The speakerphone is amplified to strengthen the sound signals that would typically be exchanged through the handset. Amplified speakers and microphones in close proximity typically cause a high-pitched whistling sound or feedback. You may have experienced public address systems doing this in cases where the volume was too high or the microphone was too close to the speakers. In order to control feedback, the speakerphone utilizes sound-activated switches. The switches monitor the microphone and the speaker allowing one or the other, but not both, to be active at any given time. This means that when you hear a caller’s voice, they cannot hear your voice, and vice versa.

8.121.3 Operation

A. From a Plexus key telephone:

Table 8 - 93 Speakerphone Operation

To perform this action	Do this
Activate speakerphone operation	Press **SPKR** or begin dialing without lifting the handset. Refer to the **On-Hook Dialing** feature.

Table 8 - 93 Speakerphone Operation

To perform this action	Do this
Deactivate speakerphone operation	Press the illuminated **SPKR** button or lift up the handset.

8.121.4 Configuration

N.A.

8.121.5 Administration

N.A.

8.122 Speed Dial: System

8.122.1 Description

The System Speed Dial feature enables up to 100 frequently dialed telephone numbers to be programmed into the system for use by all system users. Users utilize a system speed dial number by entering a feature code (Feature 55) followed by the unique two-digit speed dial code (00-99). Each programmed telephone number can be up to 20 digits in length, including pauses, *, and #.

Note
Two-second pauses may be entered into the dial string. Each pause counts as a digit for purposes of system storage capacity and the 20-digit limit.

8.122.2 Interactions

The system utilizes a line according to the user's assigned trunk Pilot-ID (see Trunk Allocation) to dial the speed dial number. If no trunk assignment is made, the speed dial will fail.

8.122.3 Operation

A. From either a Plexus key telephone or an ordinary phone:

- Enter **F55** + 2 digit Speed Dial Code.

Note
A programmable key on the keyphone may be defined as **F55**. Speed dial would then be accomplished by pressing the **F55** key and entering the 2-digit speed dial code.

8.122.4 Configuration

Each programmed number must be assigned a unique two-digit code from 00-99.

Table 8 - 94 Speed Dial: System

Method	Path	Parameter	Default
Software	System Parameters: Speed Dial	Code Number	_____ _____
Key Telephone	N.A.		

Note
When programming speed dial numbers using either Plexus Administrator or Key Telephone Configuration, the initial **0** for codes between 00-09 is dropped. This is done for display purposes only. It is necessary to enter both digits when using the associated speed dial number.

8.122.5 Administration

N.A.

8.123 Speed Dial: Personal

Plexus Key Telephone users can use programmable keys on the key telephone for their own frequently dialed telephone numbers. These Personal Speed Dial numbers are in addition to the System Speed Dial numbers accessed by entering a feature code followed by a unique two-digit speed dial code. Personal Speed Dial numbers are utilized by pressing the associated programmable key.

8.123.1 Interactions

N.A.

8.123.2 Operation

A. From a Plexus key telephone:

- Hold down the **OVERRIDE** key until **Use Headset:** is displayed.
- Press **VOL ▲** until **Define Keys** is displayed.
- Press **SPKR** (Not needed on 2.0 Key Telephones and above). Keys 01 to 20 will now be red.
- Press the desired **DSS** key. It will start to blink.
- Use the **XFER** key to scroll through the options to the Speed Dial option.
- Enter the number which will be assigned to that key.
- Press **OVERRIDE** again to save the change.

Note
This feature is only available to Plexus key telephones.

8.123.3 Configuration

The System Administrator can use Plexus Administrator to define users' keys. Users can assign their keys by entering programming mode on their key telephone.

In defining a programmable key with a Personal Speed Dial, the digits should be entered *exactly* as they would be dialed manually, including any trunk access digit. Up to 20 digits may be entered for each key.

Table 8 - 95 Speed Dial: Personal

Method	Path	Parameter	Default
Software	User: Features, Settings	Define Key… on Key-Telephone	Unassigned

Note
Two-second pauses may be entered into the dial string by entering a comma in Plexus Administrator or by pressing **REDIAL** on the key telephone.

8.123.4 Administration

Refer to the **PVT-30D/PDT-30D Telephone User's Guide** for more information.

8.124 Station Message Detail Recording (SMDR)

Also known as Call History Reporting.

8.124.1 Description

The Station Message Detail Recording (SMDR) feature records call information for each completed call on the system. The recorded information is output through the serial data port to either a serial printer or a call accounting device. The serial interface operates according to the following communications settings:

Table 8 - 96 SMDR

Setting	Value
Bits per Second	19200
Data Bits	8
Parity	None
Stop Bits	1
Flow Control	None

SMDR information is given in the following format:

Table 8 - 97 SMDR Field Format

Field Type	Description	Field Width
Call Type - Inbound [**I**] or Outbound [**O**]	**I** or **O**	1
Seperator Item		1
Record Type - Interim [**+**] or Final [**#**]	**+** or #, Interim indicates the call is still active, Final indicates the call has terminated	1
Seperator Item		1
Call Origination Date & Time Stamp {International Format}	DD/MM/YYYY HH:MM:SS (e.g., 19/08/1999 18:40:44)	19
Seperator Item		1
Trunk ID	Trunk ID of the call (e.g., 200)	5
Seperator Item		1
[Inbound Calls - Caller ID Digits] or [Outbound Calls - Dialed Digits]	Number called or called from (e.g., 5123289500)	20
Seperator Item		1
[Inbound Calls - Caller ID Name or DID Tag]	Name called from (e.g., BBS Telecom)	20

Table 8 - 97 SMDR Field Format

Field Type	Description	Field Width
Seperator Item		1
System Entity ID	Entity ID responsible for the call (e.g., 100)	5
Seperator Item		1
System Entity Name	Entity ID responsible for the call (e.g., John SMITH)	20
Seperator Item		3
Connection State (see key below)	See key below	3
Seperator Item		1
Elapsed Time	Time of that portion of the call HH:MM:SS (e.g., 01:23:45)	8
Seperator Item		1
Metering Pulse Count (if applicable)	used with metering capable ATI	5
Seperator Item		1
Account Code (if applicable)	Account Code associated with the call	12
Seperator Item		1
DID - Dialed Digits (if applicable)	DID digits sent on DID trunk card	20

Table 8 - 98 Key for Connection States

State	Definition
RNG	The inbound call on Trunk *X* was **RINGING** at Entity *Y* for the elapsed time
CON	The inbound/outbound call on Trunk *X* was **CONNECTED** to Entity *Y* for the elapsed time
QED	The inbound call on Trunk *X* was **QUEUED** at Entity *Y* for the elapsed time
HLD	The inbound/outbound call on Trunk *X* was **HELD** at Entity *Y* for the elapsed time
NAN	The inbound call on Trunk *X* encountered **NO-ANSWER** condition at Entity *Y* for the elapsed time
BSY	The inbound call on Trunk *X* encountered **BUSY** condition at Entity *Y* for the elapsed time
DND	The inbound call on Trunk *X* encountered **DO-NOT-DISTURB** condition at Entity *Y* for the elapsed time

Notes

An Entity can be a User, User-Group, Voice Processor or Auto-Attendant.

The Date & Time Stamp for a particular call is the instant that the call is originated. This stamp is maintained for all the interim records as well as the final record so that each record can be uniquely associated with a particular call especially when the stamp is considered in conjunction with the related trunk.

8.124.1.1 SMDR Output examples

An inbound call on Trunk 200 with Caller-ID was directly vectored to User 100. User 100 answer the call after 13 seconds, converses with the external party for 49 seconds and then blind transferred the call to User 101. User 101 answered the call after 8 seconds and spoke to the external party for 12 minutes and 34 seconds and then hangs-up. The SMDR output is as follows :-

I + 20/08/1999 19:44:39 201	5123674781	BBS TELECOM	100	John SMITH	RNG 00:00:13 0
I + 20/08/1999 19:44:39 201	5123674781	BBS TELECOM	100	John SMITH	CON 00:00:49 0
I + 20/08/1999 19:44:39 201	5123674781	BBS TELECOM	100	John SMITH	HLD 00:00:02 0
I + 20/08/1999 19:44:39 201	5123674781	BBS TELECOM	101	Jane BINAN	RNG 00:00:08 0
I # 20/08/1999 19:44:39 201	5123674781	BBS TELECOM	101	Jane BINAN	CON 00:12:34 0

An outbound call on Trunk 200 was initiated by User 100 which lasted for 1 minute and 27 seconds. The SMDR output is as follows :-

O # 20/08/1999 20:13:37 201	3289500	100	John SMITH	CON 00:01:27 0

8.124.2 Interactions

Internal calls are not recorded.

Call Conferences are recorded as multiple calls.

In the event of a power failure, SMDR operation may be maintained utilizing the Battery Backup: System feature.

8.124.3 Operation

SMDR output is always active on a Plexus system unless there is an active link between the Plexus system and the Plexus Administrator Software.

8.124.4 Configuration

To ensure accurate reports, the date and time must be set correctly. The system clock is set using the Update: Clock feature of Plexus Administrator (see chapter 6, Software Configuration) or through Key Telephone Configuration.

8.124.5 Administration

N.A.

8.125 Station Relocation

8.125.1 Description

Plexus systems are designed to enable convenient relocation of both users and their station equipment. Authorization Management allows users to log-in at any physical station on the system. Therefore, a user may relocate simply by logging out at one location and logging in at another. In cases where the user wants to relocate their equipment, as in the case of a key telephone with the user's preferred programmable key definitions, the equipment is simply disconnected at the former location and reconnected at the new location. If the user is not a transient user and they were assigned to the former extension port for Home Station purposes, it is also advisable to assign the user to the new extension port. Refer also to "Port Management: Line and Station" - section 8.108.

8.125.2 Interactions

N.A.

8.125.3 Operation

N.A.

8.125.4 Configuration

N.A.

8.125.5 Administration

N.A.

8.126 System Administrator Privileges (SAP)

8.126.1 Description

The maintenance and configuration activities listed below can only be performed from a station at which System Administrator Privileges (SAP) have been activated.

1. Manually change between Day and Night Service.
2. Utilize the Alarm Clock feature.
3. Perform Hotel/Motel functions. See Hospitality Package.
4. Record Auto-Attendant greetings for an Auto-Attendant Interface (AAI). Refer to the optional AAI section in the product manual for more information.
5. Activate/deactivate ports.

8.126.2 Interactions

Only users with the Administrate privilege can activate SAP.

Note
Users with administrate privilege will automatically have the privilege enabled on Log In.

8.126.3 Operation

A. From either a Plexus key telephone or an ordinary phone:

Table 8 - 99 System Administrator Privileges

To perform this action	Do this
Activate System Administrator Privileges	Enter **F91** + Password. (The default password is **123456**.)
Deactivate System Administrator Privileges	Enter **F90**.

8.126.4 Configuration

Table 8 - 100 System Administrator Privileges

Method	Path	Parameter	Default
Software	User: Privileges	Administrate	Not Granted
Key Telephone	User: User Privileges	Can Administrate	Not Granted

8.126.5 Administration

N.A.

8.127 System Clock

8.127.1 Description

Plexus systems maintain a clock for date and time display (on display model key telephones), SMDR date and time stamp, and Day/Night service transitions.

8.127.2 Interactions

N.A.

8.127.3 Operation

N.A.

8.127.4 Configuration

The system clock is set using the Update: Clock feature of Plexus Administrator (see chapter 6, Software Configuration) or through Key Telephone Configuration.

Table 8 - 101 System Clock

Method	Path	Parameter	Default
Key Telephone	System Time: Set System Time	Set Year Select Month Select Day of Month Select Weekday Select Hour Select Minute	_____ _____ _____ _____ _____ _____

8.127.5 Administration

N.A.

8.128 System Message Display

8.128.1 Description

Plexus Key Telephone users with a display model key telephone receive several informative messages from the system. These display messages present call activity and line status information including Caller ID information and calling party information for internal calls. Refer also to “Caller ID” - section 8.55 and “Busy Inquiry” - section 8.20.

8.128.2 Interactions

N.A.

8.128.3 Operation

N.A.

8.128.4 Configuration

N.A.

8.128.5 Administration

Users can choose to have either their user ID or their user name appear as the inactive (i.e., default) display. See the Plexus Key Telephone User’s Guide for further details.

8.129 Toll Restriction

8.129.1 Description

The Toll Restriction feature prevents users from placing certain outbound calls according to telephone numbers and prefixes listed in deny dialing tables. Each user has their own deny dialing table that applies at the station where they are logged in and to calls placed by the Integrated Voice Processor (IVP), such as Remote Call Forwarding, Remote Telephone Notification, and Pager Notification calls for the user's mailbox. System deny dialing tables apply at unassigned stations (i.e., a station where no user is logged in) and calls placed by the Integrated Voice Processor for voice mailboxes that are not associated with a user or are associated with a user group.

Note
Typically system deny dialing tables are extremely restrictive so as to prevent unauthorized calls from a public or unsecured station.

8.129.2 Interactions

If a user dials a number in their deny dialing table that is not specifically allowed in their allow dialing table (see "Toll Restriction Override" - section 8.130), they will hear error tone and the call will not be completed.

8.129.3 Operation

N.A.

8.129.4 Configuration

The deny dialing table lists the telephone numbers and prefixes that the system will not allow the user to dial. Telephone numbers listed in allow dialing tables specifically override any entry in the corresponding deny dialing table. For example, a user's deny dialing table may deny 1-900 numbers in general, while their allow dialing table may specifically allow 1-900-555-1212.

Table 8 - 102 Toll Restriction

Method	Path	Parameter	Default
Software	User: Deny Dialing System Parameters: Toll Restriction: Deny Dialing	Number Number	_____ _____
Key Telephone	N.A.		

8.129.5 Administration

N.A.

8.130 Toll Restriction Override

8.130.1 Description

The Toll Restriction Override feature allows users to make specific outbound calls according to telephone numbers listed in allow dialing tables. Each user has their own allow dialing table that applies at the station where they are logged in and to calls placed by the Integrated Voice Processor (IVP), such as Remote Call Forwarding, Remote Telephone Notification, and Pager Notification calls for the user's mailbox. System allow dialing tables apply to unassigned stations (i.e., a station where no user is logged in) and calls placed by the Integrated Voice Processor for voice mailboxes that are not associated with a user or are associated with a user group.

Tip

A user can override restrictions at an unassigned station by using Authorize User. See "Authorize User" - section 8.12.

8.130.2 Interactions

Allow dialing table entries override deny dialing table entries on a digit-by-digit basis according to the defined Outbound Dial Timeout. For example, if the prefix 1-900 appears in a deny dialing table and 1-900-555-1212 appears in the corresponding allow dialing table, the system will terminate the call after 1-900 unless the initial '5' is entered within the Outbound Dial Timeout period.

8.130.3 Operation

N.A.

8.130.4 Configuration

Telephone numbers listed in allow dialing tables specifically override any entry in the corresponding deny dialing table (see *Interactions* above).

Table 8 - 103 Toll Restriction Override

Method	Path	Parameter	Default
Software	User: Allow Dialing System Parameters: Toll Restriction: Allow Dialing Trunk: Features	Number Number Outbound-Dial Timeout	_____ _____ 20 seconds
Key Telephone	N.A.		

8.130.5 Administration

N.A.

8.131 Transfer Recall

8.131.1 Description

The Transfer Recall feature causes the system to recall the responsible user in the case of an unsupervised transfer to a target user that does not answer. This feature is useful in high call volume environments where it is important that callers reach the target user as opposed to a coverage entity. In such environments, Transfer Recall is enabled for the operators and receptionists that transfer the calls.

8.131.2 Interactions

If the target user is busy or in DND mode, the call will route to the applicable backup or coverage entity for the target user.

Calls that go unanswered at the target user will recall the responsible user upon expiration of the Timeout on No Answer.

8.131.3 Operation

A. From a Plexus key telephone:

- Press **DSS** Key programmed as Transfer Recall Enable/Disable (**Function 06**). The Key will be lit red if Transfer Recall is enabled or not lit if Transfer Recall is disabled.

B. From a Plexus key telephone or an Ordinary Phone:

- Enter **F81** to enable Transfer Recall
- Enter **F80** to disable Transfer Recall

8.131.4 Configuration

N.A.

8.131.5 Administration

N.A.

8.132 Trunk Allocation

8.132.1 Description

Trunk Allocation involves the creation of trunk groups and the assignment of trunks or trunk groups to users for the purpose of making outbound calls. Each user is assigned a trunk or trunk group to serve as their trunk Pilot-ID. A default trunk assignment is also made for the system. The default trunk assignment serves as the trunk Pilot-ID for unassigned stations and for certain calls placed by the Integrated Voice Processor (IVP).

Tip
When trunk groups are created, the trunks should be listed in reverse order from the hunt (i.e., rollover) order established by the phone company. Such an arrangement facilitates the use of the least busy trunks for outgoing calls.

8.132.2 Interactions

If a trunk Pilot ID is busy, a user may enable a Call Back. If the user has the appropriate privileges, he may Intrude or Monitor the trunk.

See “Call Back: External” - section 8.25, “Call Intrude” - section 8.39 and “Call Monitor” - section 8.40 for more information.

8.132.3 Operation

N.A.

8.132.4 Configuration

Table 8 - 104 Trunk Allocation

Method	Path	Parameter	Default
Software	User: Assignment: Group Access	Pilot-ID for Trunk	None
Key Telephone	User: User Attributes	Trunk Pilot-ID	None

8.132.5 Administration

Users access their trunk Pilot-ID by pressing the defined trunk access digit (see “Flexible Numbering Plan” - section 8.76).

8.133 Trunk Routing

8.133.1 Description

Trunk Routing defines the manner in which incoming calls are routed within the Plexus system. The Plexus system has the ability to route incoming calls to users, user groups, and automated attendants. Plexus systems support four different routing methods. They are Fixed, DID, Mapped DID / DNIS, and Caller ID / ANI.

8.133.1.1 Fixed

Routes the calls to the system entity defined in the trunk routing window. Trunks with fixed routing can route calls to only one system entity during the day and one system entity during the night.

8.133.1.2 DID

Direct Inward Dialing (DID) routes the calls to the system entity ID that matches the digits sent by the telephone company on a DID trunk. See chapter13, "Direct Inward Dialing (DID"

8.133.1.3 Mapped DID / DNIS

Mapped DID / DNIS routes the calls based on DID or DNIS digits received by a telephone company. The digits sent are compared to the Inbound: DID Map and routed to the defined target. See chapter13, "Direct Inward Dialing (DID)".

Note
DNIS (Dialed Number Identification Service) - The number or last few digits of the number that the caller dialed. Used with T-1/E-1 and ISDN PRI digital trunks. See chapter 10, "T-1 / E-1 CAS" and chapter 11, "ISDN PRI" for more information.

8.133.1.4 Caller ID / ANI

Caller ID / ANI routing routes calls based on received Caller ID / ANI digits from the telephone company. The digits are compared to entries in the Inbound: DID map and routed to the defined target. See chapter13, "Direct Inward Dialing (DID)".

Note
ANI (Automatic Number Identification) - The caller's telephone number. ANI sends only the telephone number of the calling party, not the name. Used by T-1/E-1 CAS. See chapter 10, "T-1 / E-1 CAS" for more information.

8.133.2 Interactions

8.133.2.1 DID Routing

The digits sent by the telephone company must exactly match the ID number of a system entity (user or user group, trunk or trunk group, or automated attendant). If the numbers do not match, the call will fail.

8.133.2.2 Mapped DID / DNIS and Caller ID / ANI Routing

The digits sent by the telephone company must exactly match entries in the Inbound: DID-Map to be routed correctly. If the digits do not match, the call will be routed to the system entity ID defined for Trunk Routing Pilot ID.

8.133.3 Operation

N.A.

8.133.4 Configuration

Method	Path	Parameter	Default
Software	Trunk: Routing: Day: Routing Method Trunk: Routing: Day: Pilot ID Trunk: Routing: Night: Routing Method Trunk: Routing: Night: Pilot ID System Parameters: Route Map: Inbound: DID-Map	Routing Method Pilot ID Routing Method Pilot ID See chapter 13, "Direct Inward Dialing"	Fixed None Fixed None
Key Telephone	N.A.		

8.133.5 Administration

N.A.

8.134 Universal Answer

8.134.1 Description

Universal Answer enables a user to answer any ringing extension or trunk on the system. In cases where multiple extensions or trunks are ringing, the user can answer a specific ringing station by entering the user or trunk ID. Refer also to “Call Pickup: Inter-Group” - section 8.43 and “Call Pickup: Intra-Group” - section 8.44.

8.134.2 Interactions

When a call is ringing to a user or trunk mapped on the user’s Key Telephone, the **DSS** or **DTS** key will begin flashing red. The Key Telephone may be set up to audibly notify the caller of these ringing calls (see “Call Notify” - section 8.41). The user may answer these calls by pressing the flashing key.

8.134.3 Operation

A. From a Plexus key telephone:

Table 8 - 105 Universal Answer

To perform this action	Do this
For a general pick-up	Enter **F43**.
For a specific pick-up	Enter **F44 +** user or trunk ID. or Press the flashing **DSS** key mapped to the ringing user or trunk.

B. From an ordinary phone:

Table 8 - 106 Universal Answer

To perform this action	Do this
For a general pick-up	Enter **F43**.
For a specific pick-up	Enter **F44 +** user or trunk ID.

8.134.4 Configuration

Table 8 - 107 Universal Answer

Method	Path	Parameter	Default
Software	User: Privileges	Univ. Answer	Not Granted
Key Telephone	User: User Privileges	Can Univ Answer	No

8.134.5 Administration

To utilize Universal Answer, users issue the appropriate feature code. To answer a specific station, the associated user or trunk ID is also entered.

A programmable key on a key telephone may be defined as **F43** or **F44**.

8.135 Universal Retrieve

8.135.1 Description

Universal Retrieve enables a Plexus Key Telephone user, with the appropriate privileges, to retrieve a call (either internal or external) from another user's Busy Hold Queue or from a user group's Busy Hold Queue.

Universal Retrieve also allows users to receive call notification of calls in user group Hold Queues. In order to be notified of these calls and retrieve these calls, the user must have a **DSS** key mapped to monitor the user group.

See "Call Notify" - section 8.41.

Note
Universal Retrieve is not available with Plexus Key Telephone firmware version 1.01 or previous versions.
The firmware version is displayed on a display model key telephone when the telephone initializes. To initialize a key telephone, dial **Feature 99**.

8.135.2 Interactions

For a user to retrieve a call from another user's Busy Hold Queue, the user must have the held party mapped on a **DSS** or **DTS** key. The user will receive no indication that the call is held and the mapped key will be solid red.

To receive notification that there is a call in the user group's Busy Hold Queue, the user must have the user group mapped on a **DSS** key.

For a user to retrieve a call from a user group's Busy Hold Queue, the user must have the user group mapped on a **DSS** key or must have the held party mapped on a **DSS** or **DTS** key.

8.135.3 Operation

8.135.3.1 To retrieve a call from a user's Busy Hold Queue:

- Press the **DTS** or **DSS** key mapped to the held call.

8.135.3.2 To retrieve a call from a user groups Busy Hold Queue:

- Press the flashing **DSS** key mapped to the user group. If multiple calls are in the user group's Busy Hold Queue, this will retrieve the call that has been on hold the longest.

or

- Press the **DTS** or **DSS** key mapped to the held call.

8.135.4 Configuration

Table 8 - 108 Universal Retrieve

Method	Path	Parameter	Default
Software	User: Privileges	Univ. Retrieve	Not Granted
Key Telephone	User: User Privileges	Univ. Retrieve	No

8.135.5 Administration

To utilize Universal Retrieve, the user must press the associated **DSS** or **DTS** key.

8.136 Unsupervised Conference

8.136.1 Description

The Unsupervised Conference feature allows a user, with the appropriate privileges, to release a call conference made up exclusively of external calls. Releasing such a conference allows the external calling parties to continue their call without a system user present. The privilege associated with Unsupervised Conference also allows a user to transfer one external call to another external call (i.e., trunk-to-trunk transfer).

8.136.2 Interactions

N.A.

8.136.3 Operation

A. From a Plexus key telephone:

- While connected to an external number, press the **CONF** key.
- Call another external number, and press the **CONF** key.
- To release external parties from the conference, press **XFER** + **F89** and hang up.

B. From an ordinary phone:

- While connected to an external number, press **FLASH**.
- Call another external number, and press the **FLASH**.
- To release external parties from the conference, **FLASH** + **F89** and hang up.

8.136.4 Configuration

The Unattend Trunks privilege must be granted for users that will need to establish unsupervised conferences or perform trunk-to-trunk transfers.

Unsupervised conferences and trunk-to-trunk transfers automatically terminate upon expiration of the Unattended Call Duration.

Table 8 - 109 Unsupervised Conference

Method	Path	Parameter	Default
Software	User: Privileges System Parameters: Feature, Timeouts	Unattend Trunks Unattended Call Duration	Not Granted 15 minutes
Key Telephone	User: User Privileges	Can Unattend Trunks	No

8.136.5 Administration

N.A.

8.137 User Alias

8.137.1 Description

Plexus system users can be assigned multiple user IDs to serve as aliases. Typically aliases are used for multitasking purposes. For example, an individual that performs sales and tech support functions in an organization can have a sales user ID (that is a member of the sales user group) and a tech support user ID (that is a member of the tech support user group). If the user signs in both user IDs they will receive both sales and tech support calls. Refer also to "Authorization Management" - section 8.11.

8.137.2 Interactions

- A unique user ID needs to be assigned for each alias.
- Several aliases (e.g., sales, technical support) may be signed in at the same station.
- An alias may only be signed in at a single station. A subsequent sign in of the alias at a second station will automatically sign the alias out at the initial station.
- Sign in and sign out require a password if one is assigned.

8.137.3 Operation

N.A.

8.137.4 Configuration

N.A.

8.137.5 Administration

The privileges for an extension are based on the user logged in at the phone. When a different user/alias logs in, the privileges will change accordingly.

8.138 Voice Mail: Fully Integrated

8.138.1 Description

The Plexus system offers fully integrated voice mail and a full-menu auto-attendant through the Plexus Integrated Voice Mail option available with an Integrated Voice Processor (IVP) peripheral card. Several configurations are available with regard to channels, storage capacity, and language. Contact your local dealer for further details.

8.138.2 Interactions

N.A.

8.138.3 Operation

N.A.

8.138.4 Configuration

N.A.

8.138.5 Administration

N.A.

8.139 Voice mail: Third party

8.139.1 Description

Plexus systems support the use of third-party voice mail adjuncts (i.e., external voice adapters) by providing programmable parameters that control the different DTMF tones that are exchanged between the system and the device for inbound signaling.

8.139.2 Interactions

N.A.

8.139.3 Operation

N.A.

8.139.4 Configuration

The external voice adapter settings are accessed in Plexus Administrator by clicking on the **SET EXTERNAL VOICE ADAPTER** button located at Extension Ports: Device. Key Telephone provides a programming menu called **Ext. Voice Mail** for these parameters.

Table 8 - 110 Voice mail: Third party

Method	Path	Parameter	Default
Software	Extension Port: Device: Voice Adapter: Access	Initial Delay	500 ms
		Direct-Access DTMF Prefix	*
		Indirect-Access DTMF Prefix on No-Answer	#
		Indirect-Access DTMF Prefix on Busy	#
	Extension Port: Device: Voice Adapter: Call Progress, Feedback DTMF Digit On	Indirect-Access DTMF Prefix on DND	#
		Disconnect	X
		Ringing	X
		No-Answer	X
		Answer	X
		Busy	X

Table 8 - 110 Voice mail: Third party

Method	Path	Parameter	Default
		DND	X
	Extension Port: Devices	Terminal Type	Telephone

8.139.5 Administration

To use this feature, the extension port should be set as External Voice Adapter.

8.140 Volume Control

8.140.1 Description

Plexus Key Telephone users can adjust the volume levels associated with the speakerphone speaker, handset, and ringer. The user can select from eight different levels available for each volume setting.

Note
Each volume setting is independently adjusted.

8.140.2 Interactions

N.A.

8.140.3 Operation

8.140.3.1 Adjust handset or speakerphone volume

A. From a Plexus key telephone:

- Press **VOL** ▲ and **VOL** ▼ when the phone is active.

If you are using the handset, the handset volume will change.

If you are using the speakerphone, the speaker volume will change.

8.140.3.2 Adjust ringing volume

A. From a Plexus key telephone:

- Press and hold down **OVERRIDE** for 2 seconds.
- Press **VOL** ▼ until the option **Ringer Volume** appears.
- Press **CONF** or **XFER** to choose a Ringer Volume.
- Press **OVERRIDE** to accept and exit.

or

- While the key telephone is ringing, press **VOL** ▲ and **VOL** ▼.

8.140.4 Configuration

Handset volume and speakerphone volume are adjusted during use. Ringer volume is adjusted while the key telephone is ringing or through programming mode on the key telephone. See the **PVT-30D/PDT-30D Telephone User's Guide** for further details.

8.140.5 Administration

Users adjust the various volume levels as needed.

8.141 Whisper Page

8.141.1 Description

Whisper Page allows a user to page a busy user through handset/ speaker without interrupting the conversation.

Note
This Feature is only available on version 2.0 and above **Digital** Key Telephones. If a user attempts this Feature on a target user with an Analog Key Telephone, the results will not be to standard or guaranteed.

8.141.2 Interactions

The target user (paged user) must be on a Digital Keyphone only. The calling user will not be able to hear the target user's conversation.

8.141.3 Operation

A. From a Plexus key telephone:

- Dial target user extension or press associated **Dss** Key.
- Upon receiving busy signal, press **FLASH + F14.**
- State message and disconnect.

or

- Dial target user extension or press associated **Dss** Key.
- Upon receiving busy signal, press **Dss** key programmed as **F14.**
- State message and disconnect.

B. From a ordinary phone:

- Dial target user extension.
- Upon receiving busy signal, **FLASH + F14.**
- State message and disconnect.

8.141.4 Configuration

Table 8 - 111 Whisper Page

Method	Path	Parameter	Default
Software	User: Privileges User: Override	Intrude ALL Intrude	Not Granted
Key Telephone	User: User Privileges	Can Intrude Call	No

8.141.5 Administration

N.A.

8.142 XOS Upgrade Management

8.142.1 Description

Each new release of the Switch Operating System (XOS) is easily and conveniently uploaded to the system using Plexus Administrator. The new XOS file may be uploaded locally through a direct serial connection or remotely through a modem connection. See chapter 6, "Software Configuration" and “Software (Windows-based) configuration” - section 8.120

8.142.2 Interactions

New XOS sometimes requires a new configuration file (.zdb). A simple software utility is provided with the XOS that will convert configuration files to ensure compatibility with the new XOS (see “Upgrade configuration using Dbconv.exe” - section 3.3.5.

Note
It will typically be necessary to configure certain new parameters in the converted file in order for the system to operate properly. It is recommended that the new configuration file at least be reviewed from within the latest release of Plexus Administrator before being uploaded to the system.

The Windows-based PC running Plexus Administrator communicates with the system through a 9-pin serial cable connected between the RS-232 port on the DXP card (the motherboard on the Micro-Plexus) and an available serial (COM) port on the PC or through a modem connection (see “Remote System Management (RSM)” - section 8.117).

8.142.3 Operation

N.A.

8.142.4 Configuration

The XOS is upgraded by right-mouse clicking on the image of the DXP card in Plexus Administrator and selecting ‘Upgrade XOS…’ after opening a link (i.e., Link: Open). See “Updating system XOS” - section 6.31 for further details.

8.142.5 Administration

N.A.

IVP Configuration

9

9.1 System Overview

9.1.1 System Overview

The Plexus Digital Telephone System offers fully integrated Voice Mail and Automated Attendant functionality with the addition of the optional Integrated Voice Processor (IVP) peripheral card. This chapter separates the components and features of the cards into the following sections:

- "The Integrated Voice Processor (IVP) Peripheral Card"
- "The Automated Attendant"
- "The Voice Mailboxes"
- "The System Parameters"
- "The Voice Mail System Manager"
- "Setting up Hold Queue Intercepts"
- "Multi-Language Option"

9.1.2 The Integrated Voice Processor (IVP) Peripheral Card

The IVP card is the physical device inserted into the system cabinet to provide the necessary functionality for the voice mail system (e.g., hard drive to store voice messages and prompts). In the case of some models of the Micro system, the Integrated Voice Processor is incorporated on the system motherboard.

The IVP card, like any other peripheral card, can be configured through either the Plexus Administrator Windows application or through a Plexus key telephone with a display panel.

9.1.3 The Automated Attendant

The Automated Attendant (a.k.a. auto-attendant) is the system component that routes calls to various Plexus system (e.g., users and user groups) and voice mail (e.g., voice mailboxes) resources. The auto-attendant tree developed by the Voice Mail System Manager determines how the auto-attendant operates. The architecture of the Plexus voice mail system is such that the auto-attendant is extremely flexible in meeting the unique call routing needs at each Plexus installation.

Note
The auto-attendant tree may be modified and maintained via the Plexus Administrator Windows application or through any DTMF-capable telephone handset (see “Voice Mail System Manager” - section 9.6).

9.1.4 The Voice Mailboxes

Voice mailboxes are the system components that provide storage for voice messages. Voice mailboxes may also perform such advanced functions as notifying pagers and remote telephone numbers when new messages have arrived. Mailboxes are typically created by the Voice Mail System Manager for each user on the system, but they may also be created for user groups, telecommuting employees, overflow, and other specialized applications.

Note
The voice mailboxes may be created and maintained via the Plexus Administrator Windows application or through any DTMF-capable telephone handset (see “Voice Mail System Manager” - section 9.6).

9.1.5 The System Parameters

System Parameters are IVP settings that affect the system. They include the Voice Mail System Manager passwords, the Silence Detect Threshold among others. These two settings can be set by the VPS Attribute Editor. The others must be modified through the Voice Mail System Manager.

9.1.6 The Voice Mail System Manager

The Voice Mail System Manager is the individual or group of individuals trained to administer the voice mail system. Administration functions include the development and maintenance of the auto-attendant tree and the creation and maintenance of voice mailboxes.

9.1.7 Setting up Hold Queue Intercepts

The IVP can be defined as the Intercept Pilot ID for user and user group Hold Queues. The intercept will periodically reassure the user that they are on hold and the call will be answered. The IVP card offers a default system message to be played or a custom message to be created for this Intercept.

9.1.8 Multi-Language Option

The Multi-Language Option allows incoming callers to select a language in which the menu options will be announced. Mailbox owners may select the language in which their mailbox options are read.

9.2 The Integrated Voice Processor (IVP) Peripheral Card

9.2.1 Inserting the IVP Peripheral Card

The IVP card is inserted into the system cabinet like any other peripheral card Refer to chapter 4, "Installation."

9.2.2 Adding an IVP Card to an Established Plexus System

When adding an IVP card to an established Plexus system, modifications need to be made to the Plexus configuration file to account for the addition of the new peripheral card.

To modify the existing configuration:

1. Launch Plexus Administrator.
2. From the **File** menu, select **Open**.
3. Select the appropriate configuration file. Plexus configuration files have an extension of **.zdb**.
4. Modify the configuration according to the instructions below for creating and configuring Voice IDs.
5. After completing the configuration changes, upload the modified configuration file according to the instructions in the **Software Configuration** section of the Plexus Product Manual.

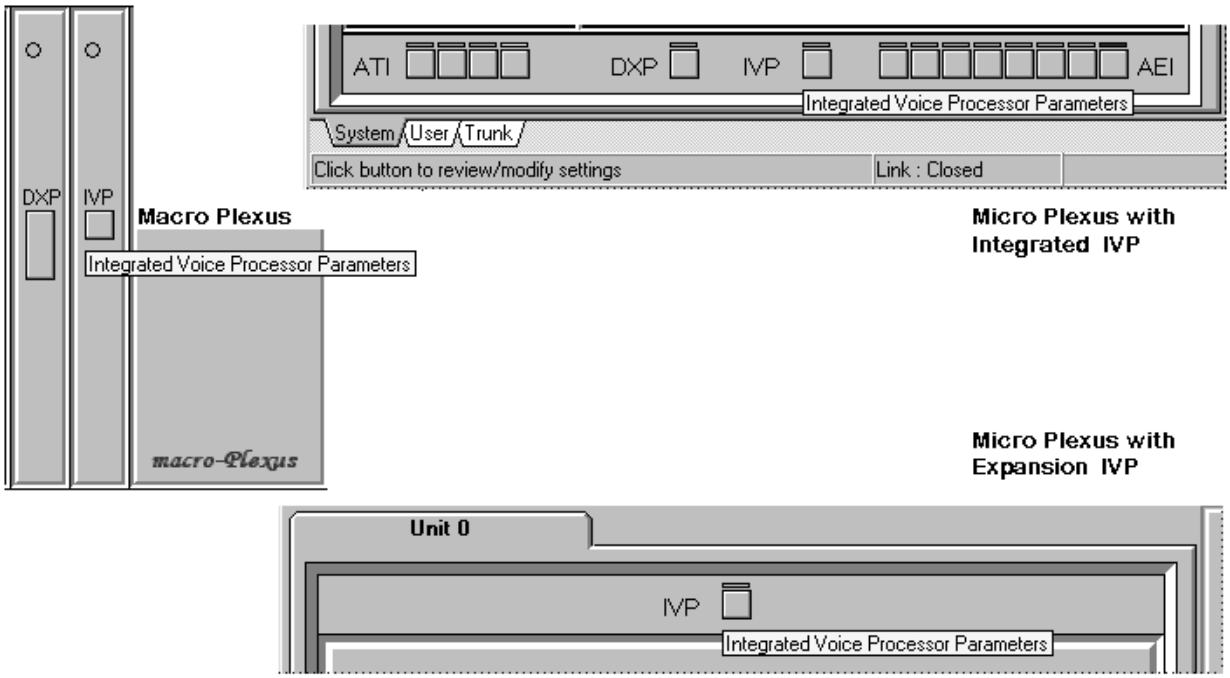
9.2.3 Incorporating an IVP Card with the Original Configuration

When incorporating an IVP card at the time of original install and configuration, follow the instructions below in conjunction with the instructions for the other system components and peripheral cards found in chapter 6, "Software Configuration."

Note
Refer to “Inserting Peripheral Cards” - section 4.3 for instructions on card layout. An image of the IVP card must appear in the proper slot before the following configuration instructions can be carried out.

1. Click on the **System** tab within Plexus Administrator to display an image of the system cabinet and the available peripheral cards.

2 Click on the button on the image of the Integrated Voice Processor (IVP) card to configure the voice processor.

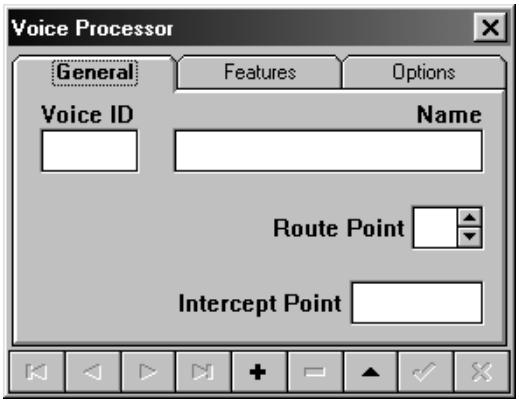


9.2.4 Configuring the IVP peripheral card

9.2.4.1 Create Voice IDs

1 Click on the button on the image of the IVP card, as shown above.

2 The following dialog window will be displayed.



3 To create a Voice ID, click on [+] and enter a Voice ID Sequence. The Voice ID sequence is the start number of the range of logical IDs assigned to voice mail system resources.

- The Voice ID sequence may begin with any number based on the **Dial-Number Range** system parameter. Refer to the **Dial Plan** tab under System Parameters in your Plexus System Manual for more information.
- Note that the voice mail system, unlike the Plexus system, only supports a **Dial-Number Range** width of 2-4.
- The default settings call for a range from 1 to 4 with a width of 3. Therefore, the Voice ID sequence may be any number between 100 and 499.

4 After entering a Voice ID Sequence, additional voice mail resources (i.e., Voice IDs) may be created by clicking on [+].

5 Existing Auto-Attendant IDs may be edited from this dialog box by selecting the Edit button: [▲].

6 After editing an Auto-Attendant ID, the Post Edit button ([✓]) must be selected to record the changes. See “Navigating in Voice Processor Properties” on page 7 for more information about the Edit and Post Edit buttons.

9.2.4.2 Navigating in Voice Processor Properties

The 9 buttons at the bottom of each Voice Processor tab scroll between existing Voice IDs, create additional Voice IDs according to the established sequence, delete existing Voice IDs, and prevent configuration errors from being posted to the system. The Edit record button must be selected prior to making any changes on a tab.

The buttons and their functions are as follows:

First Record Scrolls to the first Voice ID in the sequence.

Prior record Scrolls to the prior Voice ID.

Next record Scrolls to the next Voice ID.

Last record Scrolls to the last Voice ID in the sequence.

Insert record Inserts a new Voice ID.

Delete record Deletes an existing Voice ID.

Edit record Enables the parameters of a Voice ID to be edited.

Post Edit Posts changes made to the Voice ID parameters.

Cancel Edit Cancels changes made to the Voice ID parameters.

9.2.4.3 Configuring Voice IDs

Scroll to the desired Voice ID and make appropriate entries and selections on each of the tabs presented below.

General Properties

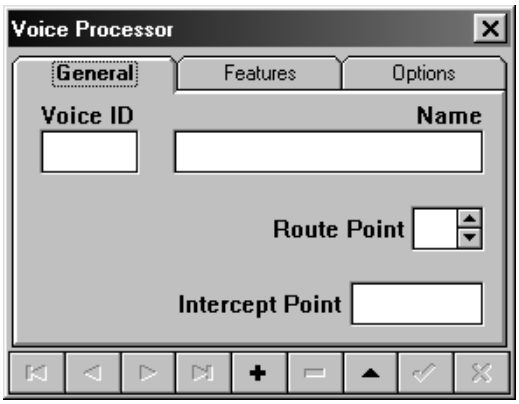


Table 9 - 1 General

Voice ID		
Logical identifier for the voice mail resource. ***Note*:** The ID is automatically assigned based on the Voice ID Sequence entered when the first Voice ID was created.	Numeric 2 to 4 digits, (depending on the dial plan)	
Name		
Informal identifier for the voice mail resource. E.g., Day Message	Alphanumeric Up to 20 characters	
Route Point		
The Route Point for the auto-attendant tree with which this Voice ID will be associated.	0-9	0 *
Intercept Point		
The main **dial node** that the Intercept is off of. For example, an Intercept may be created for a user or user group. See “Setting up Hold Queue Intercepts” - section 9.7.	Numeric	

* = default settings

Features

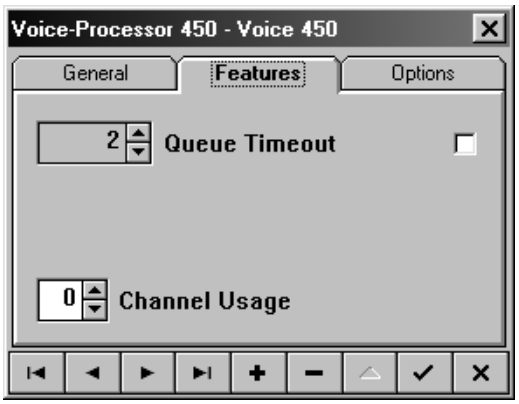
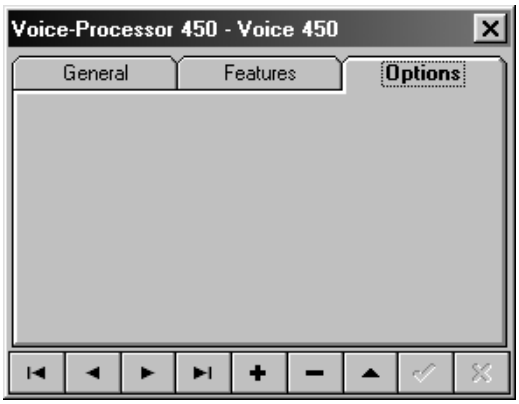


Table 9 - 2 Features

Hold Queue Timeout		
The Hold Queue Timeout refers to the period of time that a call may remain in queue, waiting for an available channel, before going to the Busy Cover-Pilot. ***Note***: The checkbox enables the Hold Queue. If the Hold Queue is not enabled (i.e., checked), calls will go straight to the Busy Cover Pilot in cases where there are no available channels.	0-255 minutes Enable/Disable*	2 minutes *
Channel Usage		
To be implemented in a future release.		

* = default settings

Options



To be implemented in a future release.

9.2.5 Key Telephone Configuration of the IVP

Refer to Chapter 7, "Administration by Keyphone" for instructions on entering programming mode and utilizing the Key Telephone Configuration template.

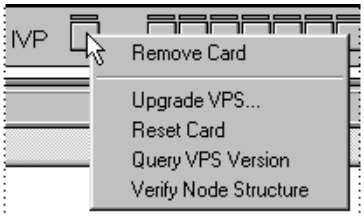
The IVP card must be physically inserted into the system cabinet before key telephone configuration can be carried out.

9.2.6 Using the IVP menu choices

There are four menu choices when right-clicking on the IVP card in an open configuration file. To access these menus, proceed as follows:

1. Launch Plexus Administrator.
2. From the **File** menu, select **Open**.
3. Select the appropriate configuration file. Plexus configuration files have an extension of **.zdb**.
4. From the **Link** menu, select **Open**.
5. The link indicator at the bottom of the screen should indicate **Link: Opened** and the LED image should appear green. (See “Establishing a link” - section 6.27 for more information).

6 Right click on the IVP port of the IVP Card.



Note
The menu choices will not be available without an active link.

7 Point to the menu choice desired and left click once.

There are four IVP menu choices available. Each provides a tool for IVP configuration or maintenance.

9.2.6.1 Upgrade VPS

Upgrade VPS is used to upgrade Node configurations, Mailbox configurations, IVP operating software, or System parameter changes. Refer to “The Automated Attendant” - section 9.3, “Voice Mailboxes” - section 9.4, “Updating IVP operating software” - section 9.2.7, or “System Parameters” - section 9.5 for more information.

9.2.6.2 Reset Card

Reset card is used to reset the IVP card without power cycling or resetting the entire system. An IVP card would have to be reset if the digit width of Voice Mailboxes has been changed or after updating the IVP operating software. Refer to “Voice Mailboxes” - section 9.4 or “Updating IVP operating software” - section 9.2.7 for more information.

9.2.6.3 Query VPS Version

Query VPS Version queries the current version of the IVP operating software. This would be done if the system is changing Plexus Administrator software versions or if directed to by BBS Telecom Technical Support.

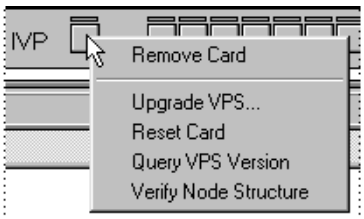
9.2.6.4 Verify Node Structure

Verify Node structure reads the node configuration description from the IVP card. This can be very helpful if the original node configuration has been lost or is not available. Once the Plexus Administrator has read the node configuration, it will save the file as **Vps-node.ndb** in the saved file of the Plexus Administrator folder. You will have to open this node configuration to view it (see "Modifying an existing node structure" - section 9.3.7 for details). The node configuration will only have the node descriptions and properties. **No recordings** will be downloaded from the IVP card, only the node descriptions and properties.

9.2.7 Updating IVP operating software

On occasion, it may be necessary to upgrade the IVP's operating software. This is usually done when upgrading to newer versions of the Plexus Administrator software or when directed to upgrade by BBS Telecom Technical Support. To upgrade the IVP operating software, proceed as follows:

1. Launch the Plexus Administrator Windows application.
2. Open the system's configuration file (**.zdb** file)
3. From the **Link** menu, select **Open**.
4. The link indicator at the bottom of the screen should indicate **Link: Opened** and the LED image should appear green. (See "Establishing a link" - section 6.27 for more information).
5. Right click on the IVP card.
6. From the pop-up menu, select "Upgrade VPS".



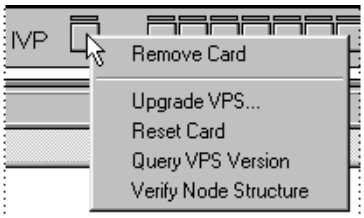
7 Locate the updated IVP operating software. You can get this from BBS Telecom Technical Support or from the BBS Telecom web site at www.bbstelecom.com. The file will be named something very similar to **orc3*xx*.vps** with the ***xx*** being the most recent version (e.g., **orc352.vps**).

8 Click on **OK**.

9 A status indicator at the bottom of the screen will display the progress of the upload. When completed, a window will popup with **VPS Upgrade Complete.** Click on **OK.**

10 Right click on the IVP card.

11 From the pop-up menu, select "Reset Card."



12 The card will take between 10-30 seconds to reset. IVP resources will not be available during this time.

Note
Upgrading the IVP operating system will not delete or reset existing node configurations or mailbox configurations already stored on the card.

9.3 The Automated Attendant

9.3.1 Description

The Integrated Voice Processor utilizes an Automated Attendant (a.k.a. auto-attendant) to route calls to various system (e.g., users and user groups) and voice mail (e.g., mailboxes) resources. The structure of the auto-attendant, referred to as the auto-attendant tree, is customized to meet the call routing needs at each Plexus installation. An auto-attendant tree is developed by creating and linking nodes. Each node performs a different function (e.g., present a menu of choices, dial an extension, access a voice mailbox, etc.).

9.3.2 Node types

9.3.2.1 Menu Node

A Menu Node plays a recorded message for callers and directs calls to subsequent nodes based on the digits (i.e., DTMF tone) entered by the callers. Also see “Menu Node Properties” - section 9.3.15.1.

Menu Nodes allow the digits 0-9 to be entered by the caller. The ***** and **#** keys are also defined within a menu node. By default, entering the ***** key followed by a mailbox number logs the caller into a mailbox (e.g., to check messages). By default, entering the **#** key followed by a mailbox number plays the associated mailbox greeting and allows the caller to leave a message in the mailbox.

The ***** and **#** keys can also be set to restart the auto-attendant or be ignored when pressed as an alternative to the default actions discussed above.

9.3.2.2 Dial Node

A Dial Node instructs the auto-attendant to dial a specific system entity (e.g., a user or user group) and route the call to that entity. Also see “Dial Node Properties” - section 9.3.15.3.

9.3.2.3 Get Extension Node

A Get Extension Node records DTMF tones entered by a caller. The tones recorded may then be passed on to a Dial Node to dial a specific system entity (e.g., user or user group). Typically, the main auto-attendant greeting includes the option to dial a party's extension. Also see "Get Extension Node Properties" - section 9.3.15.2

9.3.2.4 Mail-Box Access Node

A Mailbox-Access Node directs a call into a mailbox and plays the mailbox greeting. Mailbox-Access Nodes may be configured to allow the caller to leave a message or to simply play the greeting and then exit. Also see "Mailbox-Access Node Properties" - section 9.3.15.5.

9.3.2.5 Play Message Node

The Play Message Node is a multi-purpose node. Play Message Nodes are primarily used to play informational messages to callers, but they also may be used for the following functions:

- To record your own message.
- To play the User ID (extension) entered in a previous Get Extension Node or Directory Node.
- To direct a call to a different sub-tree of the auto-attendant.
- To play the name of the incoming caller.
- To play silence for a specified period of time.
- To play Music-On-Hold for a specified period of time.
- To play the name of the owner of the mailbox specified by the caller in a previous Get Extension Node or Directory Node.

Play Message Nodes may be utilized almost anywhere in an auto-attendant tree. They are most useful when a recorded message needs to be played to the caller, but no DTMF digits need to be collected. Also see "Play Message Node Properties" - section 9.3.15.4.

9.3.2.6 Directory node

A Directory Node announces the names and mailbox numbers (optional) of each user on the voice mail system. While the voice mail system is playing the users' names, the caller can press the "#" key to make a selection. The voice mail system can then attempt to dial the corresponding user. Also see “Directory Node Properties” - section 9.3.15.7.

9.3.2.7 Interrogate node

Interrogate Nodes are used to present a caller with a series of messages (e.g., questions), each requiring a verbal response from the caller. The voice mail system can deposit the series of responses in a mailbox as a single message. The Interrogate Node is useful for asking a caller a series of short questions such as name, address, and phone number. Also see “Interrogate Node Properties” - section 9.3.15.6.

9.3.3 Route points

The first (left-most) digit in a node number is the Route Point. The Plexus voice mail system supports up to 10 (i.e., 0-9) Route Points. Each Route Point provides a unique location for a call to enter the voice mail system. The Route Points allow the system to route incoming calls differently based on parameters such as the time of day or the trunk (CO line) on which the call was received.

The use of different Route Points facilitates the following example applications:

- Specify different auto-attendant sequences for day and night operation. Day calls could enter at Route Point 0 while night calls could enter at Route Point 1.
- Specify different auto-attendant sequences to service multiple businesses using the same Plexus system. Incoming calls on one group of trunks (CO lines) would enter at one Route Point corresponding to Company A. Incoming calls on another group of trunks would enter at a different Route Point corresponding to Company B.

9.3.4 Route point starting point

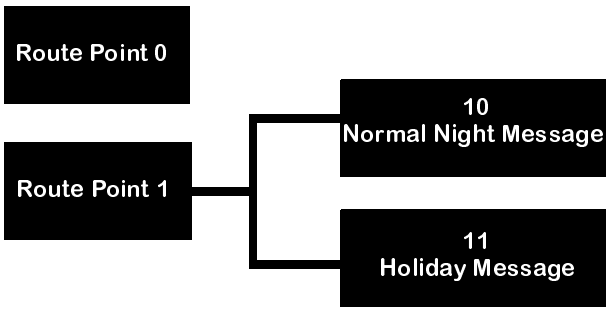
The second digit in a node number is the route point starting point. For each route point, there are 10 route point starting points (i.e., 0-9). This allows multiple auto-attendant trees to be defined for each Route Point. Only one route point starting point can be active for each route point at any given time. However, the Voice Mail System Manager can easily change the active route point starting point when logged into System Manager Mailbox #001 (see "Voice Mail System Manager" - section 9.6) or by dialing **F95**, voice ID, and the new route point starting point (e.g., dialing **F95** + 450 + 1 would change the starting point of the route point associated with voice ID 450).

Since there are 10 starting points, it is possible to have up to 10 independent auto-attendant trees defined in memory for each route point and to switch between the trees depending on current needs.

The use of different starting points facilitates the following example applications:

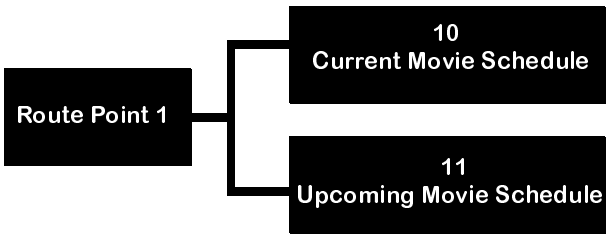
Example 1: Specify a different message for holiday weekends or office closings without deleting the normal night message

Route Point 0 is used for calls during business hours.
Route Point 1 is used for after hours call.
Route Point 1 has two associated Route Point starting points. One is for regular night calls (node 10). One is for holidays (node 11).
Each starting point (node 10 and node 11) can have a different auto-attendant tree. Alternately, they can both link to the same tree, but provide different main greetings.



Example 2: Create Route Point Starting Points to describe new products or services. A good example of this application is a movie theater.

- These messages may be recorded in advance of the release or introduction and then activated at a later date.
- Use of multiple starting points allows the movie theater personnel to record the upcoming week's movie schedule into a separate auto-attendant tree and test it completely without affecting the current schedule of movies.
- When the movies change at the end of the week, the alternate Tree is activated with the new schedule already in place.



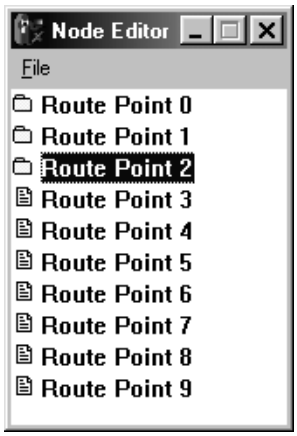
Route Point Starting Point Summary

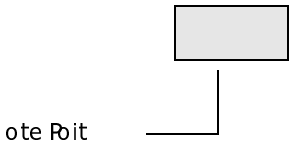
- Only one Route Point Starting Point can be active at any given time. Any calls entering Route Point 1 will route to the active auto-attendant tree.

9.3.5 Node numbering and design

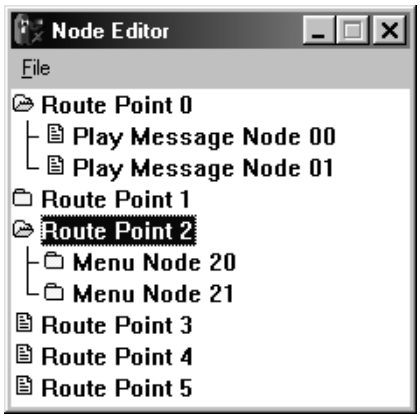
A node number is a place holder the auto attendant structure. Each node has a unique 2 - 8 digit node number, using 0 to 9 with 0 always being accessed before 1, 1 before 2, etc. The node number defines the specific place the node falls in the auto attendant.

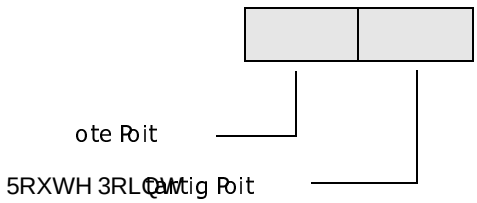
The first digit of any node signifies the Route Point that the node falls under. There are 10 available route points, 0 to 9:



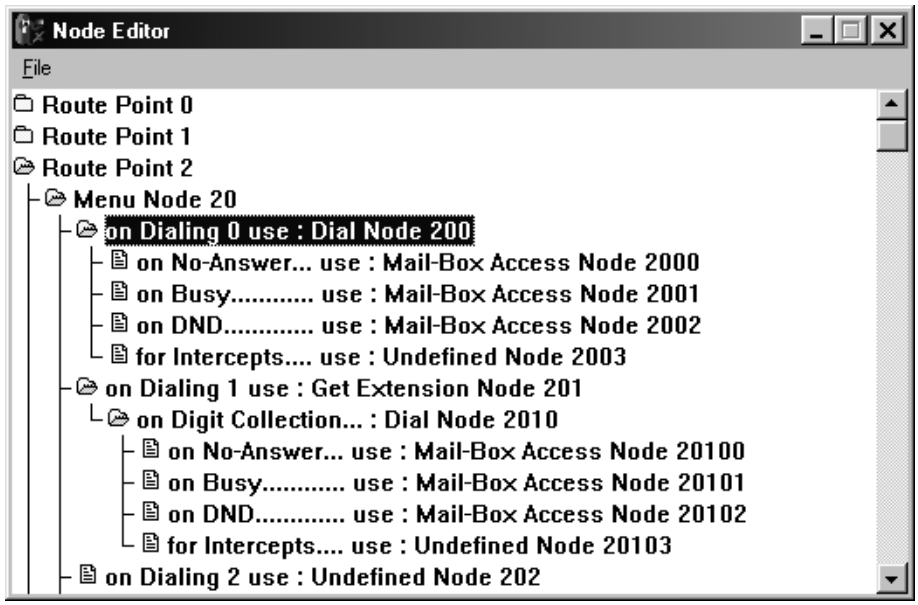


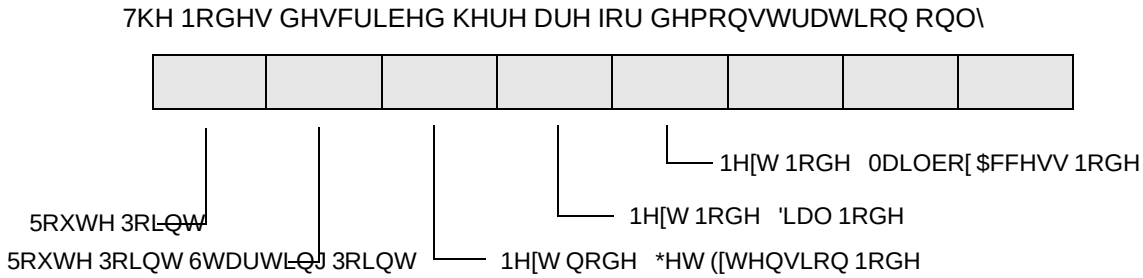
The next digit in the node represents the Route Point Starting Point. By default, the route point starts at 0. This starting point can be changed through the System Manager (see “Set the Node Starting Points (Route Point Starting Point)” - section 9.6.5.6) or by dialing **F95**, voice ID, and the new route point starting point (e.g., dialing **F95** 450 1 would change the starting point of the route point associated with voice ID 450).





The third through seventh digit (all digits except the last) represent the various node sub-trees that the node falls under.



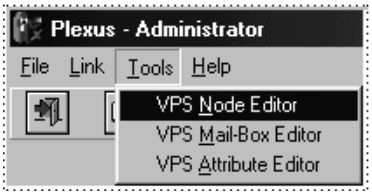


&UHDWLQJ D QHZ QRGH VWUXFWXUH

The auto-attendant tree is created using the VPS Node Editor feature of Plexus Administrator software.

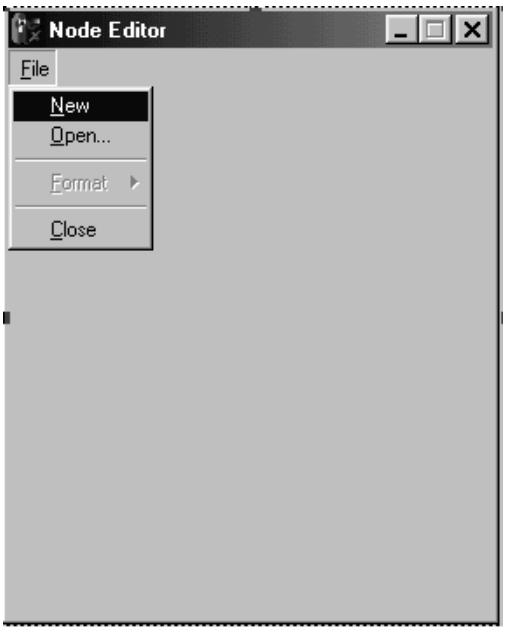
To use the VPS Node Editor:

1. Launch Plexus Administrator. Refer to the **Software Configuration** section of the Plexus Product Manual for instructions on launching Plexus Administrator.
2. From the **Tools** menu, select **VPS Node Editor**.



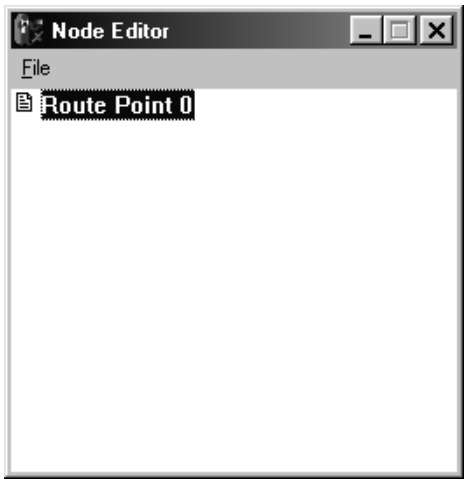
Note
The **Tools** menu will only be available if there is not a .**zdb** configuration open. If there is a configuration open in the Plexus Administrator window, close the configuration before accessing the **Tools** menu.

3 The Node Editor window will be displayed.



4 Select **New** from the **File** Menu.

5 The Node Editor dialog window will then display an initial Route Point, "0."

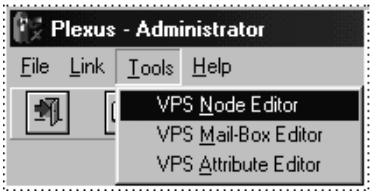


6 Nodes are added to the route point to create the node structure. To continue building the node structure, skip to “Adding a node” - section 9.3.10.

0RGLI\LQJ DQ H[LVWLQJ QRGH VWUXFWXUH

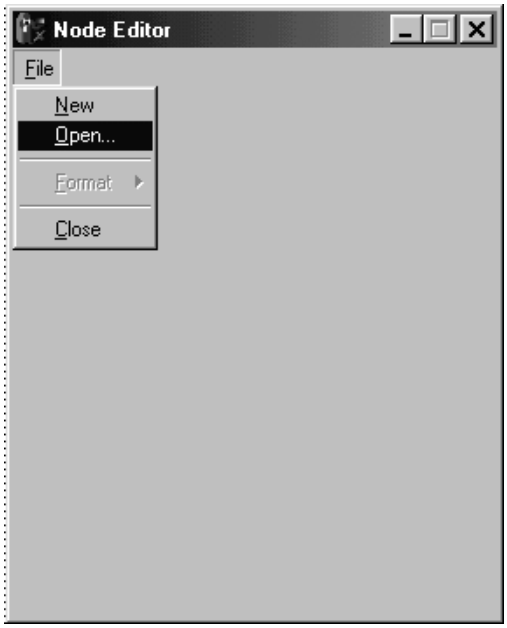
1 Launch Plexus Administrator. Refer to the **Software Configuration** section of the Plexus Product Manual for instructions on launching Plexus Administrator.

2 From the **Tools** menu, select **VPS Node Editor**.



Note
The **Tools** menu will only be available if there is not a .**zdb** configuration open. If there is a configuration open in the Plexus Administrator window, close the configuration before accessing the **Tools** menu.

3 The Node Editor window will be displayed.

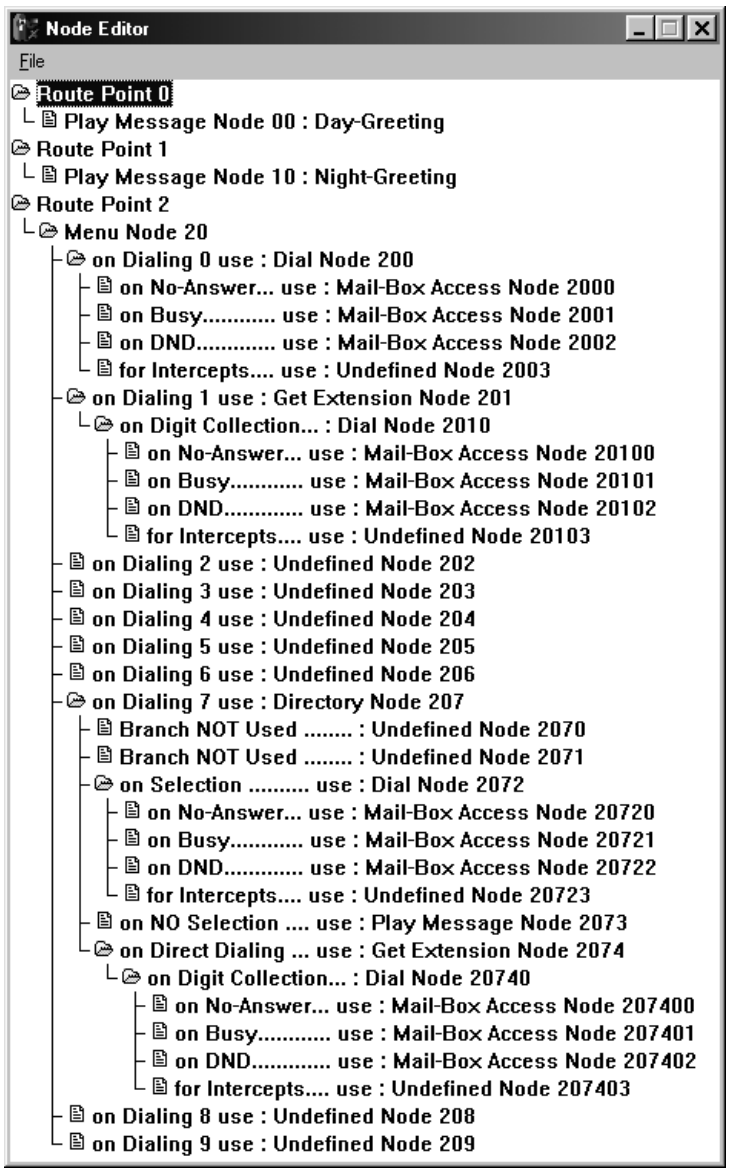


4 Select **Open** from the **File** Menu.

5 Select the desired auto-attendant configuration file. Plexus Administrator auto-attendant files will have the extension **.ndb** (e.g., **wiz-node.ndb**). Be sure to open the correct file in the correct saved folder.

6 To modify the node structure, skip to “Adding a node” - section 9.3.10.

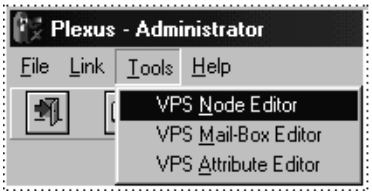
0RGLI\LQJ D :L]DUG FUHDWHG QRGH VWUXFWXUH



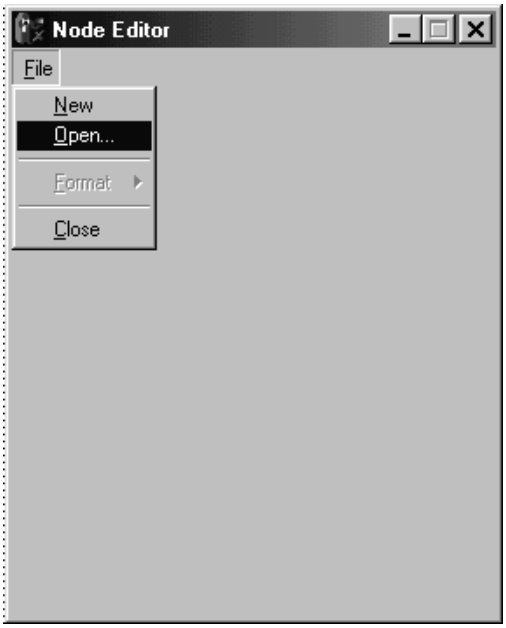
The node structure wiz-node.ndb will be created each time the Wizard creates a new configuration for a Plexus system

1 Launch Plexus Administrator. Refer to the **Software Configuration** section of the Plexus Product Manual for instructions on launching Plexus Administrator.

2 From the **Tools** menu, select **VPS Node Editor**.



3 The Node Editor window will be displayed.



4 Select **Open** from the **File** Menu.

5 Select the desired auto-attendant configuration file. Plexus Administrator Wizard auto-attendant files will have the name **wiz-node.ndb**. Be sure to open the correct file in the correct saved folder.

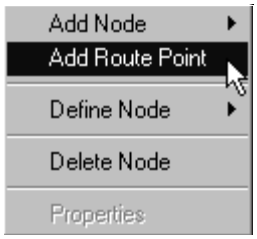
6 To modify the node structure, skip to "Adding a node" - section 9.3.10.

$GGLQJ D URXWH SRLQW

To add a Route Point to an auto-attendant tree, proceed as follows:

1 Right mouse-click anywhere in the Node Editor dialog window.

2 The following pull-down menu will be displayed.



3 Point to **Add Route Point** and click once.

$GGLQJ D QRGH

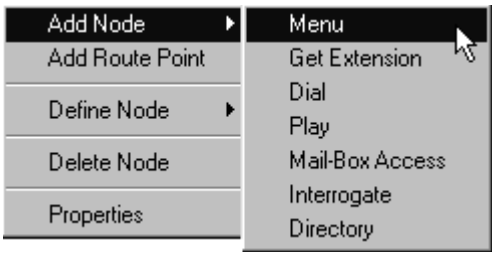
To add a node to an auto-attendant tree, proceed as follows:

1 Click once on the existing Route Point or node to which the new node should be appended.

2 The selected Route Point or node will become highlighted.

3 Right mouse-click.

4 The following pull-down menu will be displayed.



5 Point to **Add Node**.

6 Point to the desired node type on the node type pull-down menu and click once.

The addition of certain node types will automatically create other required nodes. For example, the addition of a Menu Node automatically creates an undefined node for each digit that may be dialed by a caller while in the menu node. The required nodes should be defined accordingly.

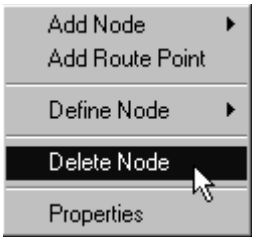
If you wish to add a node but there is already a different type node in the new nodes place, you must delete the existing node before adding the new node. To delete an existing node and the sub-nodes under that node, proceed as follows:

1 Click once on the existing Route Point or node to be deleted.

2 The selected Route Point or node will become highlighted.

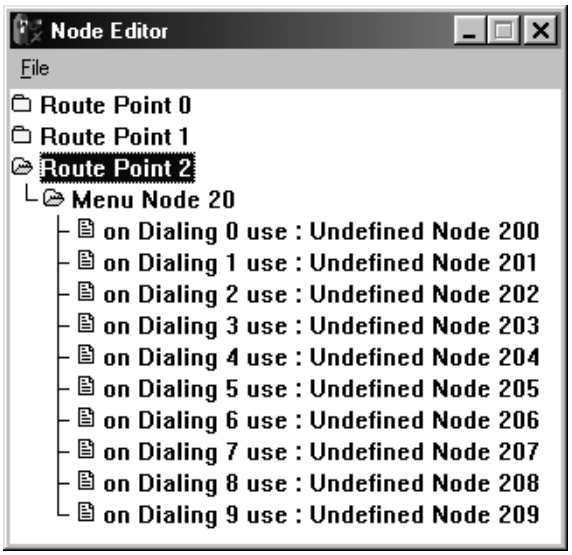
3 Right mouse-click.

4 The following pull-down menu will be displayed.

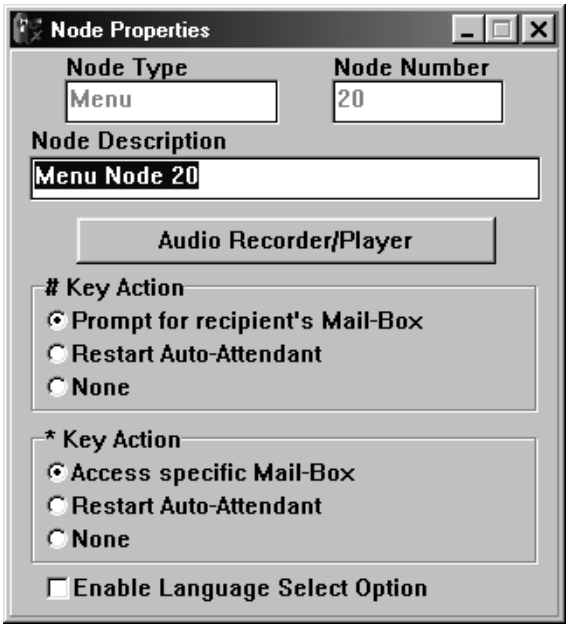


5 Point to **Delete Node** and click once.

$GGLQJ D 0HQX 1RGH



The Menu Node requires the following parameters to be set under Node Properties (see “Node Properties” - section 9.3.15):



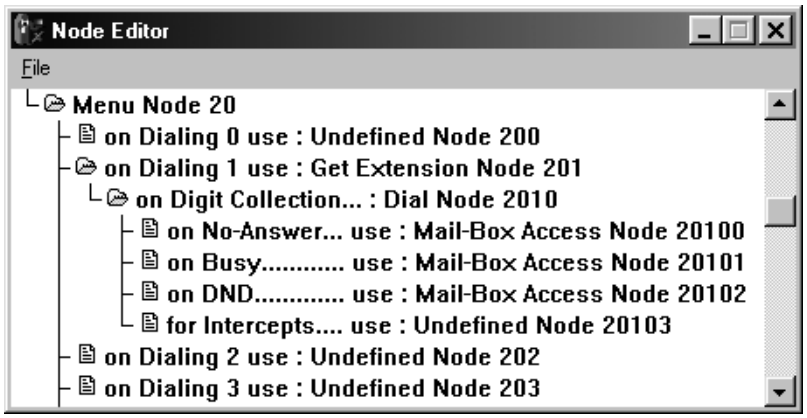
1. Audio Recorder/Player
2. # Key Action
3. * Key Action
4. Enable Language Select Option

When a Menu Node is added, the Node Editor will automatically place nodes after the Menu Node. These nodes are preset to route the call based on the dialing of 0-9 by the caller during the Menu Node Message. Each of these nodes will be undefined for Node Type. Each node will have to be defined to properly route calls.

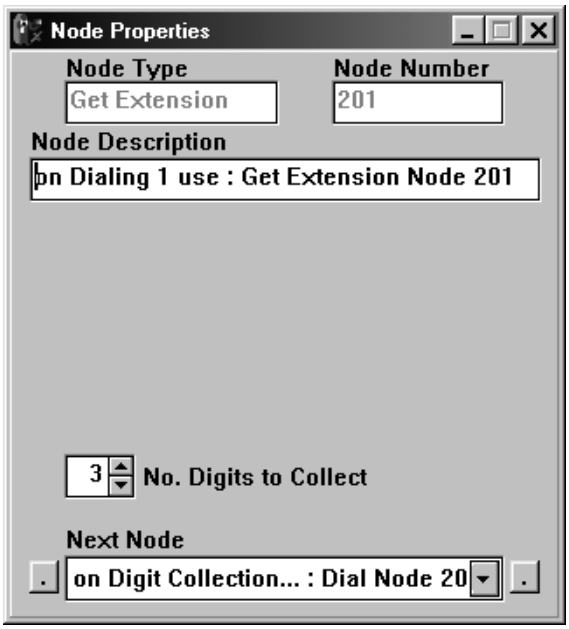
"On Dialing 0 use :" serves two purposes under a Menu Node:

1. It defines the call routing for when the caller dials 0.
2. It defines the call routing if no digit is dialed. This provides a successful call routing for callers from rotary phones or other phones where digit dialing is unavailable.

$GGLQJ D *HW ([WHQVLRQ 1RGH



The Get Extension Node requires the following parameters to be set under Node Properties (see “Node Properties” - section 9.3.15):



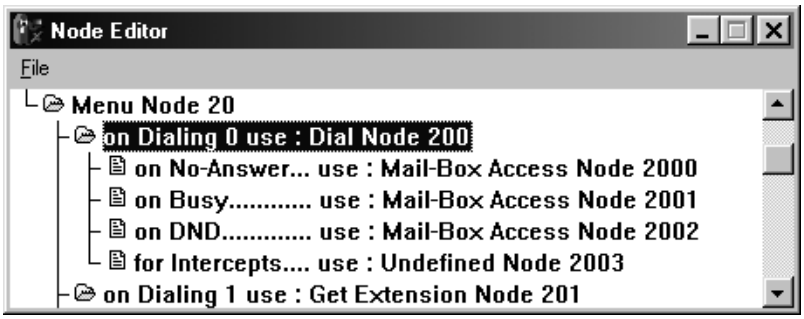
1. No. (Number) Digits to Collect
2. Next Node

When a Get Extension Node is created, five additional nodes are created by the Node Editor:

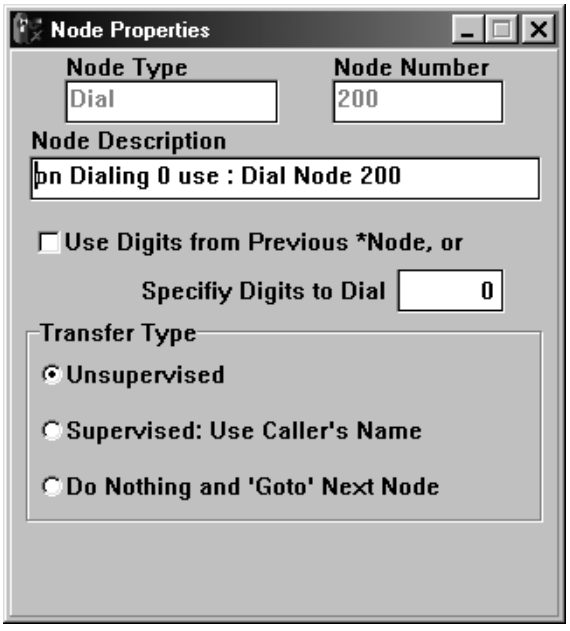
A Dial Node is created that routes the calls using the digits captured by the Get Extension Node. Mail-Box Access Nodes and a Intercept Default Node are created for call routing in cases of unavailable (no answer, busy, DND) extensions.

These additional nodes will need no modifications

$GGLQJ D 'LDO 1RGH



The Dial Node requires the following parameters to be set under Node Properties (see “Node Properties” - section 9.3.15):



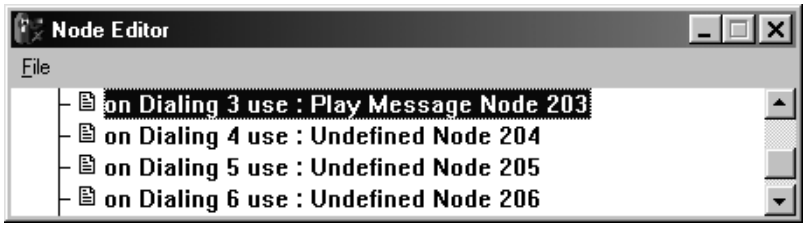
1 Use Digits from Previous *Node, or

2 Specify Digits to Dial

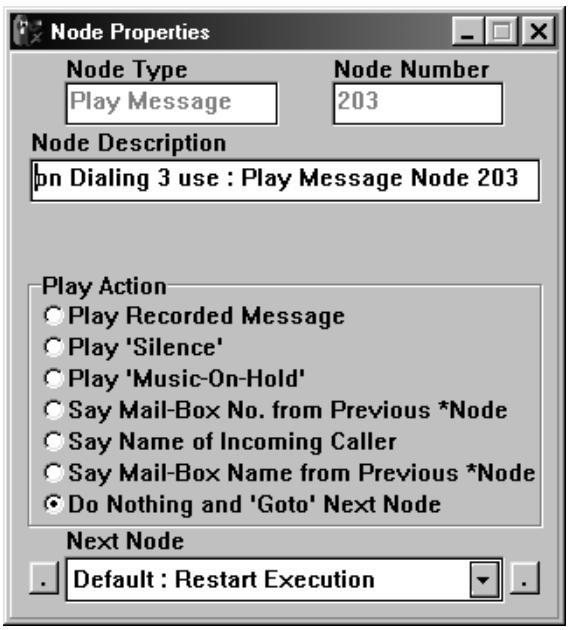
3 Transfer Type

Note
"Use Digits from Previous *Node" should only be used by Dial Nodes and Mail-Box Access Nodes when they are used as routing under a Get Extension Node or a Directory Node.

$GGLQJ D 3OD\ 0HVVDJH 1RGH



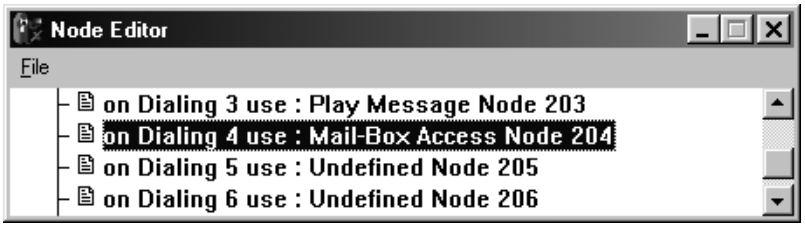
The Play Message Node requires the following parameters to be set under Node Properties (see “Node Properties” - section 9.3.15):



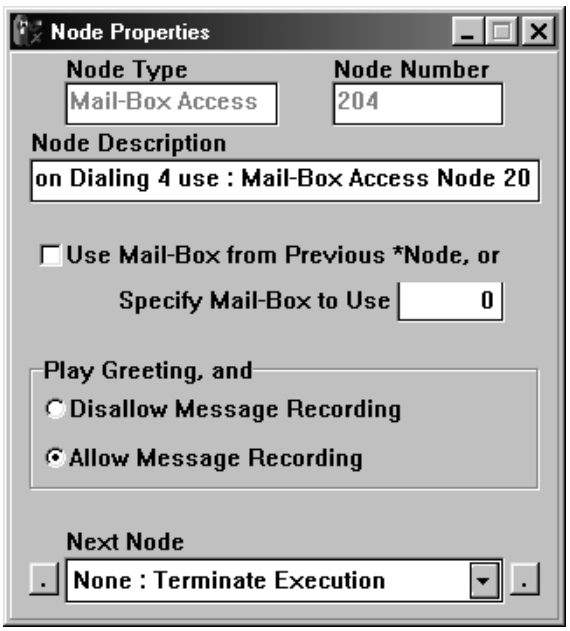
1 Play Action

2 Next Node

$GGLQJ D 0D LO %R[$FFHVV 1RGH



The Mail-Box Access Node requires the following parameters to be set under Node Properties (see “Node Properties” - section 9.3.15):



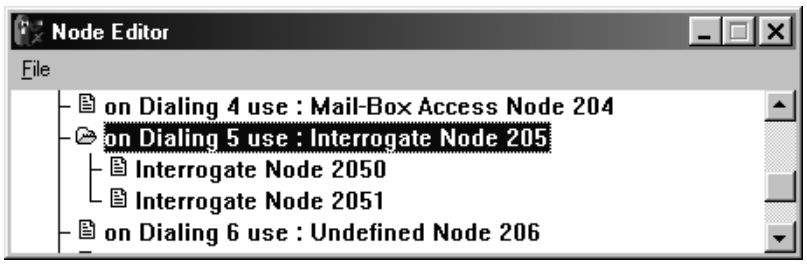
1 Use Digits from Previous *Node, or

2 Specify Mail-Box to Use

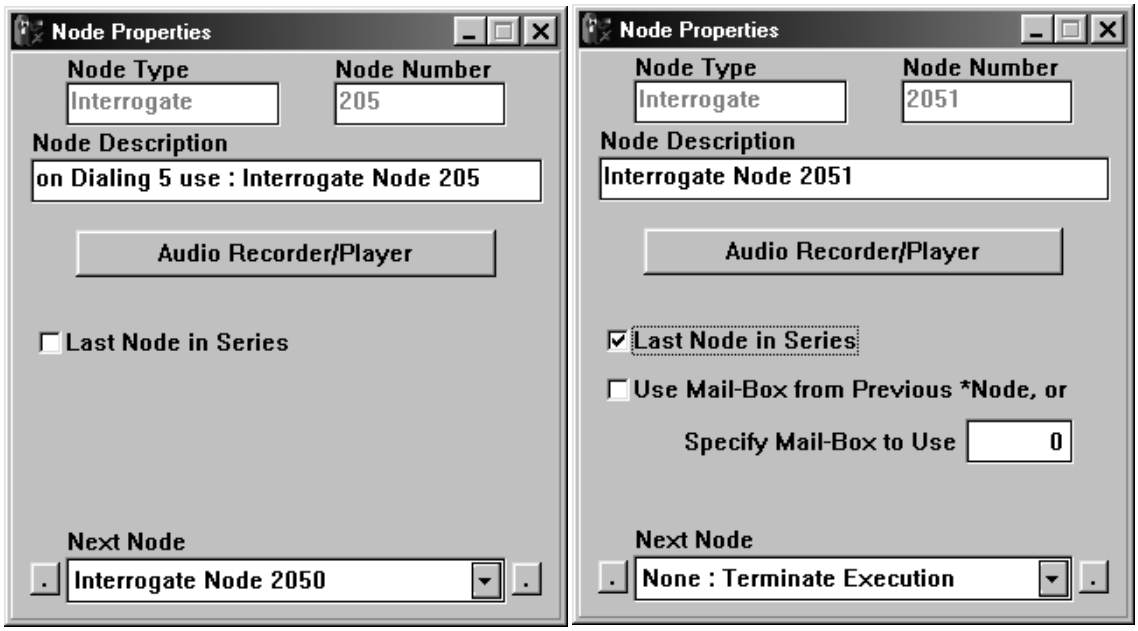
3 Play Greeting, and

4 Next Node

$GGLQJ DQ ,QWHUURJDWH 1RGH



The Interrogate Node requires the following parameters to be set under Node Properties (see “Node Properties” - section 9.3.15):



1 Audio Recorder/Player

2 Last Node in Series

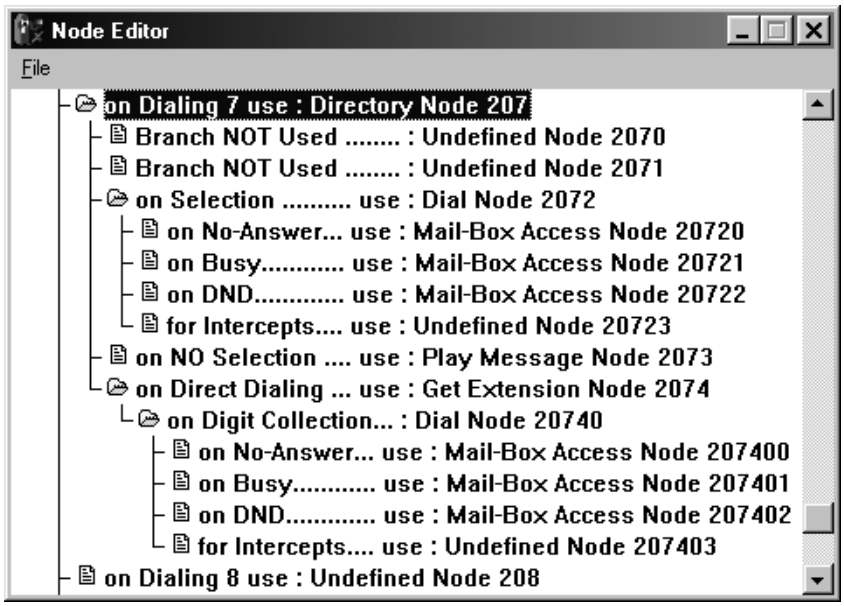
3 Use Digits from Previous *Node, or

4 Specify Mail-Box to Use

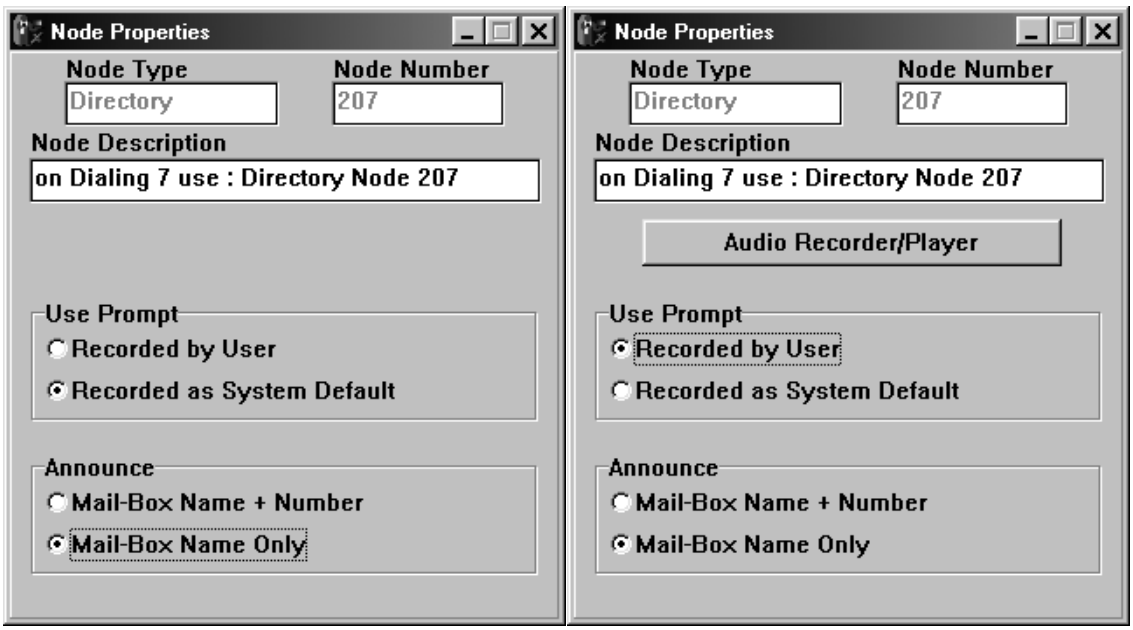
5 Next Node

Interrogate Nodes usually are used in a series to ask several questions of a caller. The last node in the series will require information as to which mailbox should receive the caller's recorded answers.

$GGLQJ D 'LUHFWRU\ 1RGH



The Directory Node requires the following parameters to be set under Node Properties (see "Node Properties" - section 9.3.15):



1. Audio Recorder/Player (if Use Prompt is "Recorded by User")
2. Use Prompt
3. Announce

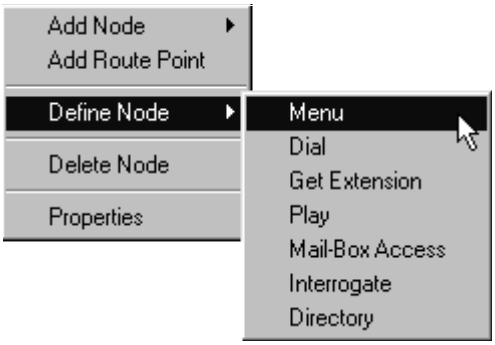
When a Directory Node is created, fourteen additional nodes are created by the Node Editor. The Node Editor will configure these additional nodes automatically, no additional configuration is required or recommended.

'HILQLQJ D QRGH

To define an existing undefined node proceed as follows:

1. Click once on the existing node.
2. The selected node will become highlighted.
3. Right mouse-click.

4 The following pull-down menu will be displayed.



5 Point to **Define Node**.

6 Point to the desired node type on the node type pull-down menu and click once.

The definition of a node as a certain node type will automatically create other required nodes. For example, the addition of a menu node will automatically create an undefined node for each digit that may be dialed by a caller while in the menu node. The required nodes should be defined accordingly.

'HOHWLQJ D QRGH

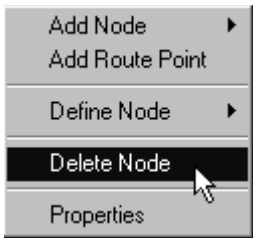
To delete an existing node, proceed as follows:

1 Click once on the existing Route Point or node to be deleted.

2 The selected Route Point or node will become highlighted.

3 Right mouse-click.

4 The following pull-down menu will be displayed.



5 Point to **Delete Node** and click once.

5HGHILQLQJ D QRGH

To redefine an existing node, it must be deleted and re-created. After deleting the node, make sure the beginning of the node tree is highlighted before adding the new node.

To redefine an existing node, proceed as follows:

1. Click once on the existing Route Point or node to be deleted.
2. Delete the node according to the directions in “Deleting a node” - section 9.3.12.
3. After deleting the node, make sure the beginning of the node tree is highlighted before adding the new node.
4. Add the new node according to the directions in “Adding a node” - section 9.3.10

0RGLI\LQJ QRGH SURSHUWLHV

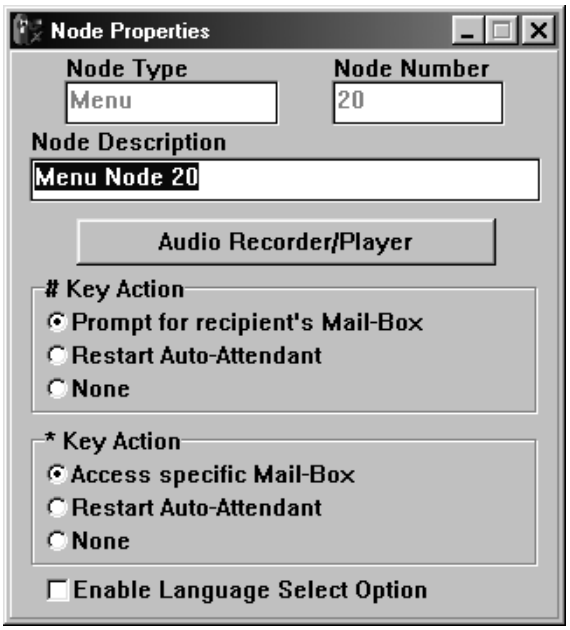
To set up the properties for an existing node, proceed as follows:

1. Click once on the existing node.
2. The selected node will become highlighted.
3. Right mouse-click.
4. The following pull-down menu will be displayed.

5. Point to **Properties** and click once.
6. The corresponding Node Properties dialog window will be displayed. Make appropriate changes on the Node Properties dialog for each node.

1RGH 3URSHUWLHV

0HQX 1RGH 3URSHUWLHV



7DEOH 1RGH 3URSHUWLHV 0HQX 1RGH

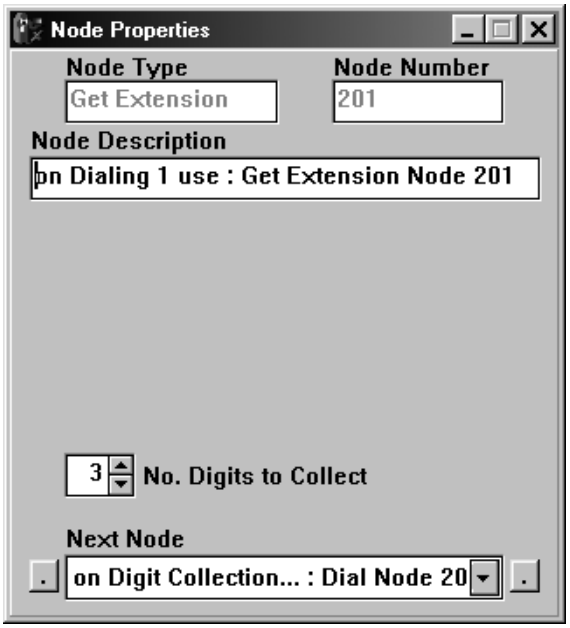
1RGH 7\SH	
The node type of the currently selected node. E.g., Menu	Automatically Identified
1RGH 1XPEHU	
Logical identifier for the node. Note: Refer to Node Numbering & Design.	Automatically Assigned
1RGH 'HVFULSWLRQ	
Informal identifier for the node. E.g., Main Greeting.	Alphanumeric Up to 30 characters
$XGLR 5HFRUGHU 3OD\HU	
Launches the Plexus audio utility for recording the menu greeting.	

7DEOH 1RGH 3URSHUWLHV 0HQX 1RGH

.
Clears the associated audio recording
.H\ $FWLRQ
Action taken when a caller presses '#' while in the Menu Node. — Prompt for a Mail-Box* Restart Auto-Attendant None
‘ .H\ $FWLRQ
Action taken when a caller presses '‘ ’ while in the Menu Node. — Access Mail-Box* Restart Auto-Attendant None
0XOWL /DQJXDJH 2SWLRQ
Enables caller to select a language from the menu. The Multi Language selection only works from the primary menu and must be directed to sub menus. Possible actions include: Press '1' for the primary language Press '2' for the secondary language Press '3' for the tertiary language The 1,2, and 3 should be directed to a corresponding submenu. **Note:** Use of this option requires the optional Multi-Language Voice Mail upgrade.

* =default settings

*HW ([WHQVLRQ 1RGH 3URSHUWLHV



7DEOH 1RGH 3URSHUWLHV *HW ([WHQVLRQ 1RGH

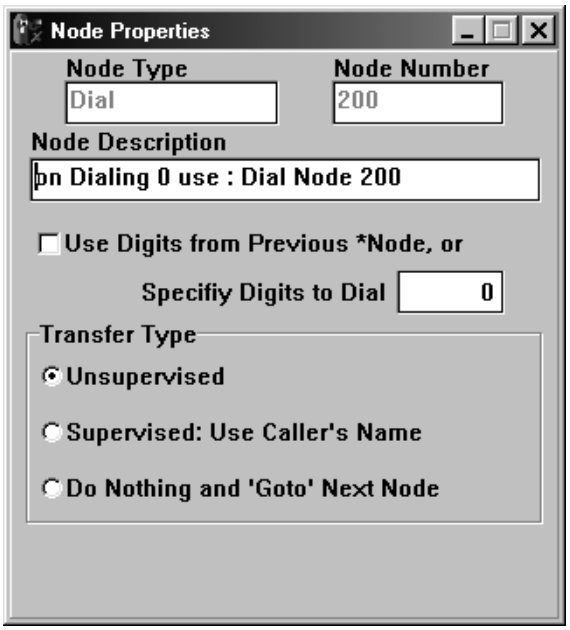
1RGH 7\SH	
The node type of the currently selected node. E.g., Get Extension	Automatically Identified
1RGH 1XPEHU	
Logical identifier for the node. Note: Refer to Node Numbering & Design.	Automatically Assigned
1RGH 'HVFULSWLRQ	
Informal identifier for the node. E.g., Collect User ID.	Alphanumeric Up to 30 characters

7DEOH 1RGH 3URSHUWLHV *HW ([WHQVLRQ 1RGH

1R 'LJLWV WR &ROOHFW		
Indicates the number of DTMF digits that the Get Extension Node should collect. When using a Get Extension Node to collect digits to dial a user or user group, this value should correspond to the width of the Dial-Number Range. Refer to the “Software Configuration” chapter of the Plexus System Manual (specifically, the Dial Plan tab under System Parameters). If the width of numbers changes from its default of 3 to either 2 or 4, the Plexus system must be powered down completely, and turned back on with the IVP card inserted. Refer to the Plexus System Manual for more information.	0-4	3*
1H[W 1RGH		
The node number of the next node to be accessed. ***Note*:** The next node will typically be a Dial Node, and will be pre-defined.	predefined by Node Editor*	
.		
left button = None: Terminate Execution right button = Default: Restart Execution		

* =default settings

'LDO 1RGH 3URSHUWLHV



7DEOH 1RGH 3URSHUWLHV 'LDO 1RGH

1RGH 7\SH	
The node type of the currently selected node. E.g., Dial	Automatically Identified
1RGH 1XPEHU	
Logical identifier for the node. **Note:** Refer to Node Numbering & Design.	Automatically Assigned
1RGH 'HVFULSWLRQ	
Informal identifier for the node. E.g., Dial user or user group.	Alphanumeric Up to 30 characters
8VH 'LJLWV IURP 3UHYLRXV 1RGH	
When selected, this parameter instructs the auto-attendant to dial the digits from a previous Get Extension or Directory Node.	Select/Deselect
6SHFLI\ 'LJLWV WR 'LDO	

7DEOH 1RGH 3URSHUWLHV 'LDO 1RGH

If Use Digits from Previous *Node is <u>not</u> selected, digits (e.g., User/User Group ID) are entered here. **Note:** This field is used in cases where the Dial Node will always connect to the same user or user group.	Up to 4 digits	0*
7UDQVIHU 7\SH		
The type of transfer that will be implemented in attempting to connect to the dialed system entity (e.g., user/user group). ***Note*:** If Unsupervised is selected and the user or user group is unavailable, the call will be routed according to the user's/user group's coverage. Refer to chapter 6, "Software Configuration"	Unsupervised (Blind)* Supervised (Transfer with Screening) Do Nothing and 'Goto' Next Node (for future release)	

* = default settings

3OD\ 0HVVDJH 1RGH 3URSHUWLHV

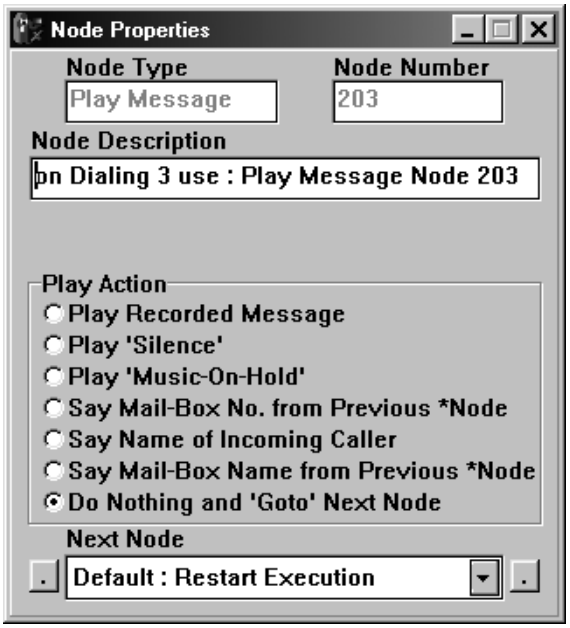


Table 9 - 6 Node Properties: Play Message Node

The node type of the currently selected node. E.g., Play Message	Automatically Identified
Logical identifier for the node. **Note:** Refer to Node Numbering & Design.	Automatically Assigned
Informal identifier for the node. E.g., Play Silence.	Alphanumeric Up to 30 characters

Table 9 - 6 Node Properties: Play Message Node

Play Action	
The action that the Play Message Node will carry out. ***Note*:** If Play Recorded Message is selected, the Audio Recorder/Player button will appear. If either Play 'Silence' or Play 'Music-On-Hold' is selected, an additional parameter, representing the duration (in seconds) of the silence or the Music-On-Hold, will appear.	Play Recorded Message Play 'Silence' Play 'Music-On-Hold' Say Mail-Box No. from *Previous Node Say Name of Incoming Caller Say Mail-Box Name from *Previous Node Do Nothing and 'Goto' Next Node*
Audio Recorder/Player	
Launches the Plexus audio utility for recording messages.	
.	
Clears audio recording associated with that node.	
Next Node	
The node number of the next node to be accessed.	Default: Restart Execution*
.	
left button = None: Terminate Execution right button = Default: Restart Execution	

* = default settings

9.3.15.5 Mailbox-Access Node Properties

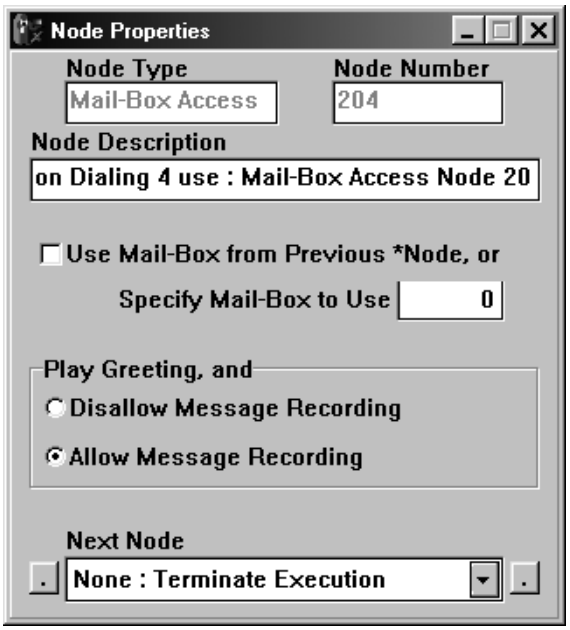


Table 9 - 7 Node Properties: Mail-Box Access Node

Node Type	
The node type of the currently selected node. E.g., Mailbox-Access	Automatically Identified
Node Number	
Logical identifier for the node. **Note:** Refer to Node Numbering & Design.	Automatically Assigned
Node Description	
Informal identifier for the node. E.g., Access the selected mailbox.	Alphanumeric Up to 30 characters
Use Mail-Box from Previous *Node, or	
When selected, this parameter instructs the auto-attendant to dial the digits from a previous Get Extension or Directory Node.	Select/Deselect
Specify Mail-Box to Use	

Table 9 - 7 Node Properties: Mail-Box Access Node

If Use Mail-Box from Previous *Node is not selected, the mailbox number is entered here. **Note:** This field is used in cases where the Mailbox-Access Node will always connect to the same mailbox.	Up to 4 digits	0*
Play Greeting		
Establishes whether the caller, upon accessing the mailbox, will be able to leave a message. ***Note***: If Disallow Message Recording is selected the mailbox greeting will be played, but the caller will be unable to leave a message in the mailbox.	Disallow Message Recording Allow Message Recording*	
Next Node		
The node number of the next node to be accessed.	None: Terminate Execution*	
.		
left button = None: Terminate Execution right button = Default Restart Execution		

* = default settings

9.3.15.6 Interrogate Node Properties

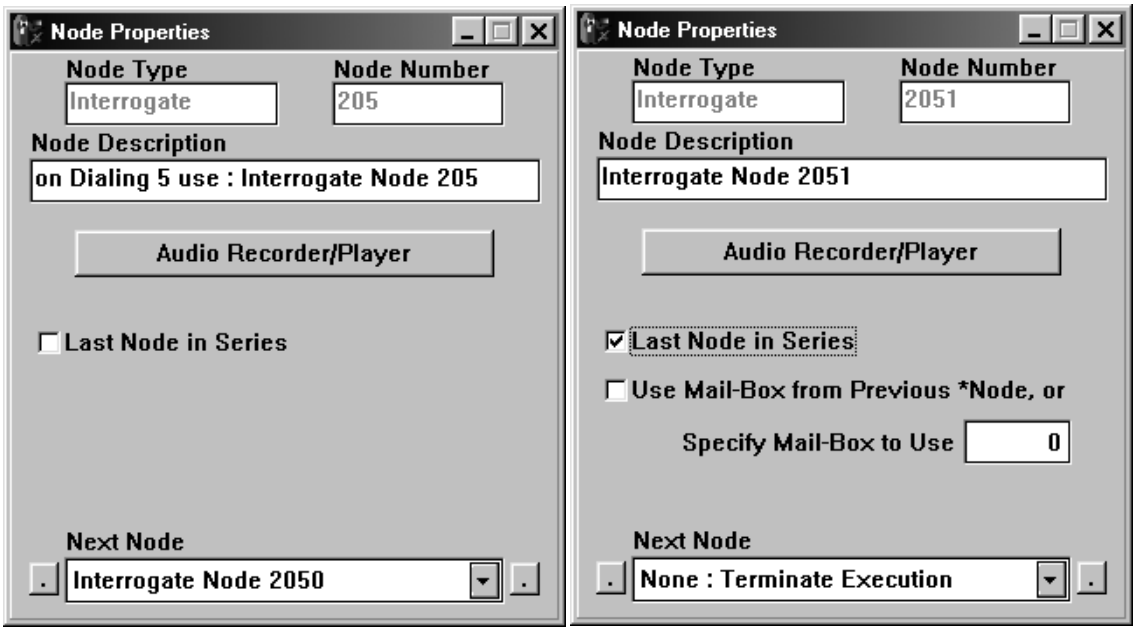


Table 9 - 8 Node Properties: Interrogate Node

Node Type	
The node type of the currently selected node. E.g., Interrogate	Automatically Identified
Node Number	
Logical identifier for the node. **Note:** Refer to Node Numbering & Design.	Automatically Assigned
Node Description	
Informal identifier for the node. E.g., Intro and First Question.	Alphanumeric Up to 30 characters
Audio Recorder/Player	
Launches the Plexus audio utility for recording messages or questions.	
Last Node in Series	

Table 9 - 8 Node Properties: Interrogate Node

Indicates that the Interrogate Node is the last node in a series of Interrogate Nodes.	Select/Deselect
Use Mail-Box from Previous *Node, or	
When selected, this parameter instructs the auto-attendant to place the interrogate responses in the mailbox indicated in a previous Get Extension or Directory Node. Note: This parameter is available only if the Interrogate Node is the last in the series.	Select/Deselect
Specify Mail-Box to Use	
If Use Mail-Box from Previous *Node is not selected, the mailbox number is entered here. **Note:** This field is used in cases where the interrogate responses will always be placed in the same mailbox.	Up to 4 digits 0*
Next Node	
The node number of the next node to be accessed.	None: Terminate Execution*
.	
left button = None: Terminate Execution right button = Default Restart Execution	

* = default settings

9.3.15.7 Directory Node Properties

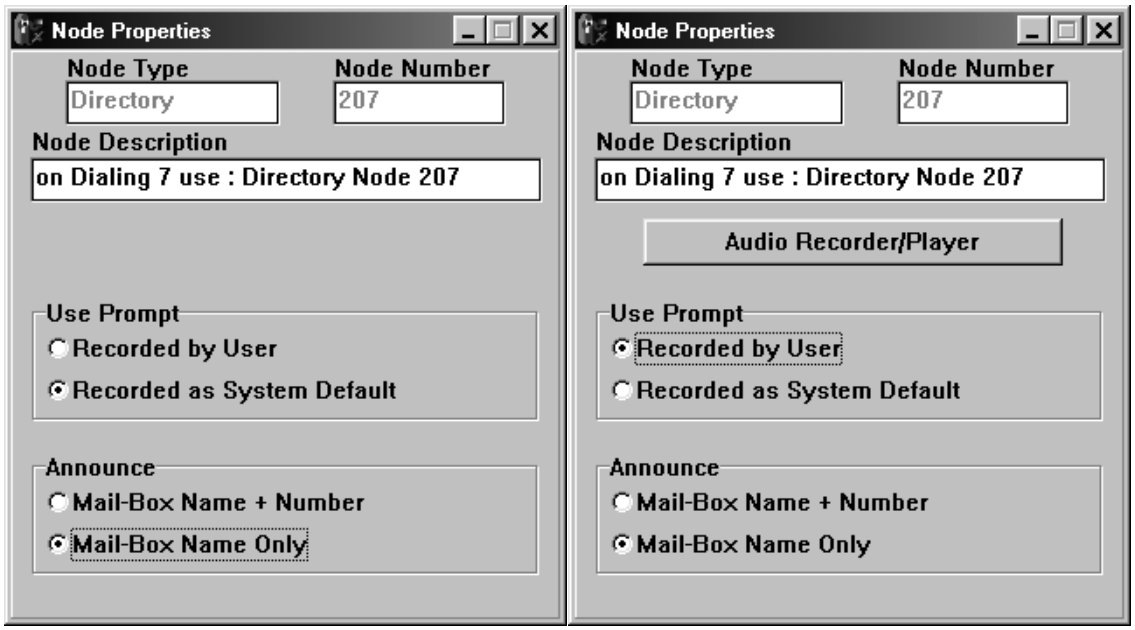


Table 9 - 9 Node Properties: Directory Node

Node Type	
The node type of the currently selected node. E.g., Directory	Automatically Identified
Node Number	
Logical identifier for the node. **Note:** Refer to Node Numbering & Design.	Automatically Assigned
Node Description	
Informal identifier for the node. E.g., Company Directory.	Alphanumeric Up to 30 characters
Use Prompt	

Table 9 - 9 Node Properties: Directory Node

Establishes whether a prompt will be recorded or if the System Default prompt will be accepted for the Directory Node. Refer to the Directory Node section for a transcript of the System Default prompt. ***Note*:** If Recorded by User is selected, the Audio Recorder/Player button will appear.	Recorded by User Recorded as System Default*
Audio Recorder/Player	
Launches the Plexus audio utility for recording the prompt.	
Announce	
Establishes what is announced to a caller in a Directory Node.	Mail-Box Name + Number Mail-Box Name Only*

* = default settings

9.3.16 Recording auto attendant messages and greetings

Several node types require an audio recording (e.g., the greeting in a Menu Node). Prerecorded .wav files may be incorporated into the node properties and utilized by the voice mail system. Alternately, recordings may be made using the Plexus Administrator audio utility.

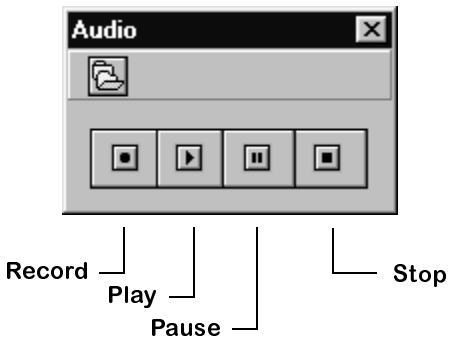
In order to use the integrated audio utility, the PC must have the following components properly configured for use within Windows:

- sound card
- speakers
- microphone

To make a recording using the integrated audio utility, proceed as follows:

1. Click on

in the Node Properties dialog window to launch the audio utility.
2. The following window will open.



3. Click on the **Record** button.
4. Speak into the microphone.
5. Click again on **Record** or click on **Stop** when complete.
6. The sound recording will be automatically saved and indexed as part of the auto-attendant configuration file.

The integrated audio utility produces **PCM 8KHz 16 Bit Mono** .wav files. Prerecorded .wav files must be converted to this format in order to be utilized in the voice mail system.

To incorporate a previously recorded .wav file, proceed as follows:

1. Click on in the Audio dialog window.
2. Select the desired .wav file.
3. Click on OK.

9.3.17 Saving the node structure

To save an auto-attendant tree developed using the Plexus Administrator **VPS Node Editor**:

9.3.17.1 For first time saves or to rename the file:

1. From the **File** menu, select **Save As**.
2. Name the file.

9.3.17.2 To save changes to a previously saved file:

1. From the **File** menu, select **Save**.

Note
Auto-attendant trees are saved as auto-attendant configuration files. Plexus Administrator auto-attendant configuration files have an extension of **.ndb**.

9.3.18 Formatting nodes

Before a completed auto-attendant tree can be uploaded to the Plexus system, the auto-attendant configuration file must be converted to .vps file format. This is accomplished via the **Format** menu option.

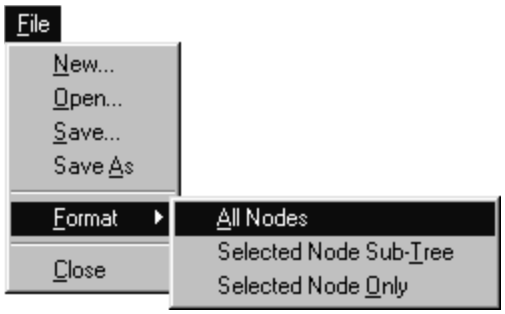
Note
The Node Editor and the Mail-Box editor both format files using .vps file format. If you format your auto attendant structure and your mailbox structure with the same name, one file will overwrite the other. Be sure to give unique names to each .vps file.

The **Format** menu option may be used to format either an entire auto-attendant tree or only certain portions of an auto-attendant tree. For example, it may be used to format a sub-tree or a node. This feature enables quick changes to be made to an already established auto-attendant tree.

Note
Formatting a single node or a node sub tree will only affect those formatted nodes. Other nodes already uploaded to the Plexus system will not be affected.

To format either an entire auto-attendant tree or a portion thereof, proceed as follows:

1. Save the auto-attendant tree or open an existing auto-attendant configuration file.
2. If only a sub-tree or a node will be formatted, select the desired portion by clicking once on the sub-tree folder or node.
3. The selected portion will become highlighted.
4. From the **File** menu, select **Format**.
5. The following pull-down menu will be displayed.



6. Point to **All Nodes**, **Selected Node Sub-Tree**, or **Selected Node Only** and click once.
7. Enter a name for the new .vps file and click on **OK**.

9.3.19 Uploading nodes

9.3.19.1 Upload the file

1. Launch the Plexus Administrator Windows application.

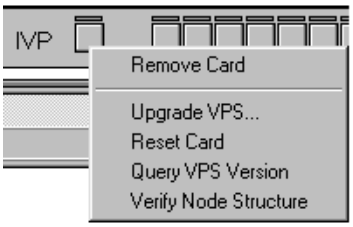
2 Open the system's configuration file (**.zdb** file)

3 From the **Link** menu, select **Open**.

4 The link indicator at the bottom of the screen should indicate **Link: Opened** and the LED image should appear green. (See "Establishing a link" - section 6.27 for more information).

5 Right click on the IVP card.

6 From the pop-up menu, select "Upgrade VPS".



7 Locate the auto attendant .vps file.

8 Click on **OK**.

9 It may take several minutes for the upload to complete. A status indicator at the bottom of the screen will display the progress of the upload. When completed, a window will popup with **VPS Upgrade Complete.** Click on **OK**.

9.4 Voice Mailboxes

9.4.1 Description

Plexus voice mailboxes serve as repositories for voice messages and other recordings and perform such advanced features as notifying pagers and remote telephone numbers when new messages have arrived. The Plexus voice mail system supports up to 511 voice mailboxes. Mailboxes are typically assigned to users, user groups, and off-site employees.

Each mailbox can store up to 128 messages and each message may be up to 6.5 minutes in length. Standard IVP peripheral cards provide over 40 hours of voice mail system storage capacity.

The mailbox number is the logical ID for the mailbox. It can be 2-4 digits in length, but it must correspond to the width of the established system Dial-Number Range (see chapter 6, "Software Configuration" for more information). The mailbox number is determined at the time the mailbox is created and it may not be subsequently modified. Typically, mailbox numbers are set to the same value as the user's User ID or the user group's User Group ID. Refer to "Importing Mailbox Users" - section 9.4.5 for more information.

Note
If the width of the mailbox numbers is changed from the default of 3 to either 2 or 4, the Plexus system must be shut down and turned back on (i.e., power-cycled) with the IVP card inserted.

9.4.2 Mailbox types

9.4.2.1 Ordinary

An ordinary mailbox is the standard single user mailbox (as opposed to a group delivery mailbox). The user has full access to the features allowed for this mailbox in the set-up. Ordinary mailboxes are generally used for users, virtual users, or play message mailboxes that do not allow for the recording of messages.

9.4.2.2 Dispatch

There are two types of group delivery mailboxes. The first type of group delivery mailbox is a dispatch mailbox and performs a “dispatch” function. A message left in a single mailbox is copied into a group of mailboxes, but only the first mailbox owner that listens to the message retains it. Other copies of the message are then removed from the other mailboxes.

Mailboxes associated with a dispatch mailbox can be notified in groups with a time delay between delivery to each group. The configuration of these dispatch groups may accomplished in two ways:

- To configure through Plexus Administrator VPS Mailbox Editor, refer to “Create a new mailbox” - section 9.4.8.
- To configure by logging into the user side of the dispatch mailbox and going to Group Settings refer to the **Voice Mail User’s Guide**.

Dispatch group 0 is notified immediately. Dispatch groups 1 through 9 are then notified according to the specified time delay.

9.4.2.3 Broadcast

The second type of group delivery mailbox is a broadcast mailbox and copies a message left in a single mailbox into a group of other mailboxes. All mailboxes then have their own personal copy of the message and can treat the message as an ordinary message.

9.4.2.4 Simple

Simple mailboxes are limited feature, individual mailboxes. Users only have access to listen to, save, and delete voice messages. Users have no access to any other mailbox features. Simple mailboxes are ideal for use with Hotel & Motel applications (see chapter 18, "Hotel / Motel Package").

9.4.2.5 Overflow

Overflow mailboxes are ordinary or group delivery mailboxes that provide a backup system to record messages if a user’s mailbox becomes full. The overflow mailboxes can be ordinary or group delivery mailboxes. The overflow mailbox must be mailbox number 999. Additional overflow mailboxes may be created in case the first mailbox becomes full. These additional overflow mailboxes must be descending sequential from 999 (998, 997, 996, ...).

If any or several mailbox(es) become full of messages (i.e., reach maximum capacity), the voice mail system will check to see if mailbox 999 exists. If it does exist, any new messages to full mailboxes will be stored in mailbox 999. If mailbox 999 does not exist, messages for full mailboxes will not be recorded.

Note
Overflow mailboxes must be created by the Voice Mail System Manager before any overflow messages will be recorded.

Note
No notification will be given of messages in the overflow mailboxes unless the mailbox is set as a group delivery mailbox. If the mailbox is to be an ordinary mailbox, it is recommended that a user be tasked to check for new messages on a regular basis.

9.4.3 Mailbox features

9.4.3.1 Associated Extension

This parameter specifies whether an extension (i.e., user) is associated with a mailbox. An extension is associated with a mailbox by default.

Note
Mailboxes which belong to off-site employees will not have an associated extension if the employee does not have a User ID.

All mailboxes assigned to users or user groups on the system should have an associated extension (i.e., user or user group). Associating an extension with a mailbox assures that the voice mail system can correctly dial the user or user group (e.g., from a Dial Node).

If a Dial Node attempts to dial '100' in a case where both a user #100 and a mailbox #100 exist, but are not associated, the call will go directly to mailbox #100.

9.4.3.2 Private Mailbox

If a mailbox is configured to be private, information about the mailbox will not be presented in a Directory Node.

Mailboxes are not private by default.

9.4.3.3 Summary of New Messages

This feature enables a mailbox to play a summary of all the new messages left in all the existing mailboxes on the system. Note that the message content remains confidential; the summary only indicates which mailboxes have new messages and how many.

The summary of new messages feature is disabled by default.

9.4.3.4 Digital Pager Notification

This feature enables a mailbox to notify a digital pager each time a new message is received. The mailbox owner specifies the pager number to be dialed. Refer to the **Voice Mail User's Guide** for more information.

The digital pager notification feature is disabled by default.

Note
A mailbox may only have Digital Pager Notification **or** Remote Telephone Notification active at the same time. If both features are active at the same time, they will cancel each other out and no notification will be sent.

9.4.3.5 Remote Telephone Notification

This feature enables a mailbox to notify a user at a remote telephone number each time a new message is received. Upon reaching the user, the voice mail system indicates that the user has messages and allows them to log into their mailbox to retrieve the messages. The mailbox owner specifies the remote telephone number to be dialed. Refer to the **Voice Mail User's Guide** for more information.

The remote telephone notification feature is disabled by default.

Note
A mailbox may only have Digital Pager Notification **or** Remote Telephone Notification active at the same time. If both features are active at the same time, they will cancel each other out and no notification will be sent.

9.4.3.6 User's Name

When configuring a mailbox, the Voice Mail System Manager can record the mailbox owner's name. This recording may be utilized by both Directory and Play Message Nodes. The mailbox owner can change this recording through user settings when they log into their mailbox [Refer to the **Voice Mail User's Guide**].

9.4.3.7 First Three Letters of the Mailbox Owner's Last Name

When using telephone configuration to configure a mailbox, the Voice Mail System Manager will need to enter the three numeric digits associated with the first three letters of the mailbox owner's name. This entry is used to identify the user in a Directory Node.

Software configuration automatically assigns these digits based on the name entered or imported into the Mail-Box Name field. Refer to "Modifying system level mailbox settings" - section 9.4.9.

9.4.3.8 Maximum Message Age

This parameter determines the number of days that a message may remain in a mailbox before it is automatically deleted.

Note
There is also a system-wide value for message age that may be configured through the telephone configuration system parameters option (option '3' under the System Manager Main Menu). The system value serves only as the initial value for mailboxes. Modifications made at the mailbox level override the system value.

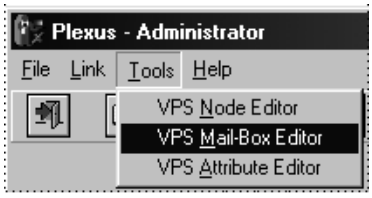
9.4.3.9 Password

Each mailbox has a password of up to four digits in length, that must be used by the owner in order to retrieve messages. The default password for every mailbox is no password. The mailbox password may be modified by the mailbox owner through user settings (refer to the **User's Guide**). The Voice Mail System Manager can reset the password to the default (e.g., in cases where a user changed the password from the default and then forgot his/her password).

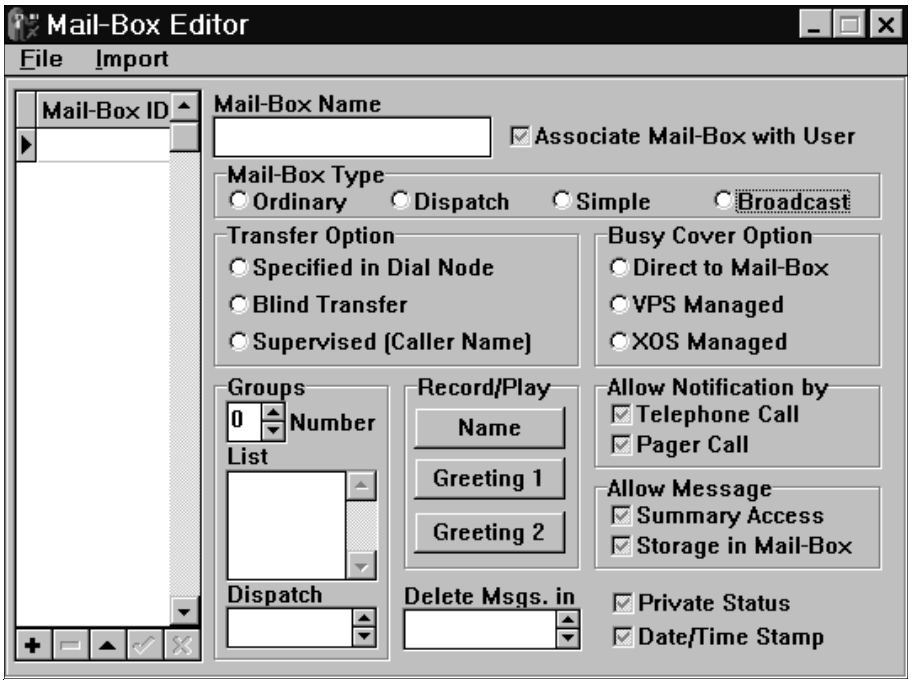
9.4.4 Creating a New Mailbox Configuration

To use the VPS Mailbox Editor:

1. Launch Plexus Administrator. Refer to chapter 6, "Software Configuration" for instructions on launching Plexus Administrator.
2. From the **Tools** menu, select **VPS Mail-Box Editor**.



3 The Mail-Box Editor dialog window will be displayed with default settings and no listed Mail-Box IDs.



9.4.5 Importing Mailbox Users

The quickest way to develop mailboxes for all users on a Plexus system is to import the user information from a Plexus configuration file (.zdb). When the import option is utilized, all users on the system are imported into the mailbox configuration and a mailbox is set up with the default parameters for each user.

To import mailbox users from a Plexus configuration file:

1 Open the VPS Mail Editor (see above for assistance).

2 From the **Import** menu, select **Mail-Box Users**

3 Select the desired Plexus configuration file. Plexus configuration files have an extension of **.zdb**.

9.4.6 Editing an Existing Mailbox Configuration

To edit an existing mailbox configuration:

1 Open the VPS Mail Editor (see above for assistance).

2 From the **File** menu, select **Open**.

Select the desired mailbox configuration file. Plexus Administrator mailbox configuration files have an extension of **.xdb**.

Note
To modify a mailbox configuration created by the Configuration Wizard, open the file **wiz-mail.xdb** under the folder created for the system you are modifying.

9.4.7 Navigating the Mailbox Configuration Window

The 5 buttons at the bottom of the Mail-Box Editor dialog window enable the creation, deletion, and editing of mailbox records. The Edit record button must be selected prior to making any changes to a mailbox. The buttons and their function are as follows:

Insert record- Inserts a new Voice ID.

Delete record- Deletes an existing Voice ID.

Edit record- Enables the parameters of a Voice ID to be edited.

Post Edit- Posts changes made to the Voice ID parameters.

Cancel Edit- Cancels changes made to the Voice ID parameters.

9.4.8 Create a new mailbox

To create a mailbox, proceed as follows.

1 Place the cursor in the Mail-Box ID list.

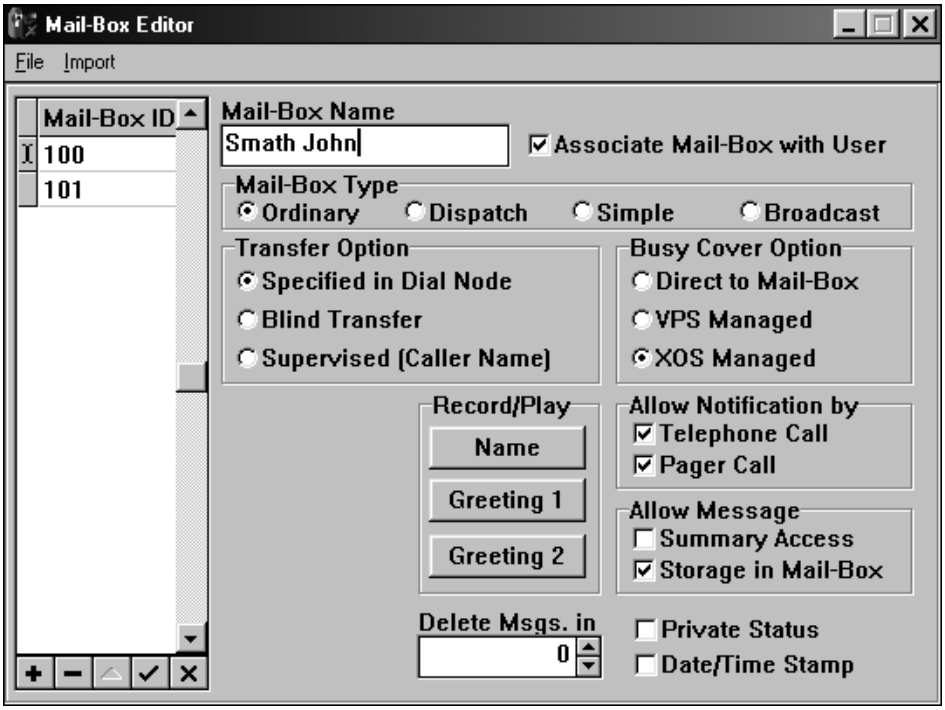
2 Click on .

3 Enter a mailbox ID (i.e., mailbox number).

4 The '}' in the Mail-Box ID list will change to a '*' to indicate that a new ID may be inserted.

5 Click on .

9.4.9 Modifying system level mailbox settings



To configure the parameters of a mailbox, proceed as follows:

1 Click on ▲ .

2 Make the desired changes to the various default parameters. Each parameter is listed in the table below.

3 Click on ✓ .

Note
To alleviate the repetitive tasks associated with configuring multiple mailboxes, right-click on a setting and select **Replicate All** or **Replicate Selected** [Replicate Selected is only available if mailboxes have been selected]. The Replicate options automatically assign a setting to all or selected mailboxes.

Note
<Ctrl> and **<Shift>** may be used to select multiple mailboxes. Click on the first mailbox so that it becomes highlighted. Hold down **<Ctrl>** and select additional mailboxes. Hold down **<Shift>** and use **<Ç >** and **<Ë >** to select a consecutive list of mailboxes.

Table 9 - 10 Modifying system level mailbox settings

Mail-Box Name	
Mailbox owner's name. Note: If the mailbox users are imported [See Import Mailbox Users], the information from the Name field on the General tab under User will automatically be entered here. ***NOTE:*** Numbers corresponding to the first 3 letters in this field will be used to identify this mailbox in a Directory Node. e.g., ABC=2, DEF=3, etc.	Alphanumeric Up to 20 characters
Associate Mail-Box with User	
Associates this mailbox with a user or user group (i.e., extension) on the Plexus system. ***Note:*** In order to associate a mailbox with a user or user group, the Mailbox ID (i.e., mailbox number) and the User or User Group ID must be the same value.	Yes*/No
Mail-Box Type	
Determines the mailbox type for the mailbox. Each mailbox type performs different functions. Note: Dispatch and Broadcast are the two different types of Group Delivery mailbox.	Ordinary* Dispatch Broadcast Simple
Transfer Option+	
Accepts or modifies the transfer method defined by the Voice Mail System Manager in the auto-attendant tree (e.g., Dial Nodes). Dial Node Specified: Accept the transfer method defined in the Dial Node. Blind Transfer: Always utilize blind transfers when transferring calls to the user. Supervised (Caller Name): Instructs the system to always ask for and record the caller's name. The caller's name will then be played for the user and the user will have the option to accept the call, send the call to his/her mailbox, or place the call on hold.	Dial Node Specified* Blind Transfer Supervised (Caller Name)

Table 9 - 10 Modifying system level mailbox settings

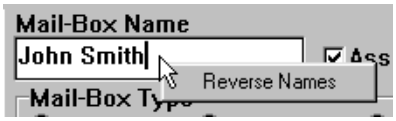
Busy Cover Option+	
Determines how a call is handled when the user or user group is busy. Direct to Mail-Box: Immediately forces callers into the mailbox owner's mailbox when the user or user group is busy. VPS Managed: Allows custom messages to be played to a caller while the call is on hold. XOS Managed: Gives control of the held call to the Plexus system and follows the established call coverage options [Refer to the Coverage tab under User and User Group].	Direct to Mail-Box VPS Managed XOS Managed*
Groups+	
Up to ten groups (0-9) may be set up for purposes of Group Delivery if this mailbox is a Dispatch mailbox.	
List+	
The members of a Group Delivery group may be defined by entering the Mailbox IDs here. Scroll to the desired group number (See Groups above) by clicking on 5and 6. Enter the Mailbox ID (i.e., mailbox number) in the List box. Press <Enter>. Note: Repeat steps 2 and 3 to add additional Mailbox IDs.	
Dispatch+	
In the case of Dispatch Group Delivery mailboxes, there is a time delay between delivery to each defined group. Group 0 is sent the message immediately. The remaining groups are sent the message according to a defined schedule. Scroll to group 1 by clicking on 5and 6. Enter the desired time delay (in seconds). ***Note***: Scroll to each subsequent group and repeat steps 1 and 2. No delay time is necessary for group 0. E.g., If group 1 is set up for a 10-second time delay and group 2 is set up for a 20-second time delay, group 1 will get dispatch messages 10 seconds after group 0 and group 2 will get dispatch messages 30 seconds after group 0. ***Note****:* Placing a user in more than one dispatch group will cause the user to receive multiple notifications of a new message and receive inaccurate notification of the number of new messages. Once a dispatch group has received notification of a new message, the members of the group will retain the message notification even if additional dispatch groups are notified.	
Record/Play+	

Table 9 - 10 Modifying system level mailbox settings

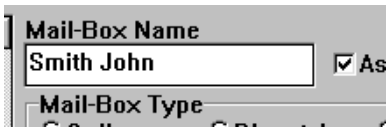
The mailbox owner's name and the two available greetings are recorded by clicking on the corresponding buttons [See Recording Mailbox Greetings]. ***Note**:* The mailbox owner may change such default recordings. However, mailbox owner changes will be overwritten with subsequent mailbox configuration uploads [See Formatting a Mailbox Configuration File].	
Enable Notification by Telephone Call	
Enables the mailbox to notify a user at a remote telephone number each time a new message is received.	Enable/Disable*
Enable Notification by Pager Call	
Enables the mailbox to notify a digital pager each time a new message is received.	Enable/Disable*
Enable Message Summary	
Enables the mailbox to play a summary of all the new messages left in all the existing mailboxes on the system.	Enable/Disable*
Enable Message Storage+	
Enables callers to leave messages in the mailbox.	Enable*/Disable
Private Status	
Designates the mailbox as a private mailbox for purposes of directory nodes.	Select/Deselect*
Date/Time Stamp+	
Instructs the mailbox to play the date and time stamp when playing messages. Note: The date and time stamp indicates the date and time that a message was received.	Select/Deselect*
Delete Messages In	
Designates the number of days a message may be saved in the mailbox before it is deleted. ***Note:*** 0 days = never	# days 0*

+= May be modified by the mailbox owner.
* =default settings

If a directory node is going to be created with a custom message asking for the first three letters of the persons first name, the names in the Mail-box Name field can be reversed by



All mailboxes will have the names reversed.



9.4.10 Modifying user level mailbox settings

Several setting can only be modified from the user level or can be overridden by the user. These setting are accessed by logging into a mailbox and pressing **3** to "to make changes to your mailbox settings".

To make changes to user level mailbox settings:

1. Press **MSG** key
2. Enter * + Mailbox ID to make changes to
3. Enter the mailbox password (if applicable, default is no password)
4. Press **3**
5. Press **8** the number of times indicated below to access each menu item listed.

Table 9 - 11 Modifying user level mailbox settings

Press '8" X times to skip to this option	Option	Parameters
0	Language Selection	English Spanish Korean French Portuguese

Table 9 - 11 Modifying user level mailbox settings

Press '8" X times to skip to this option	Option	Parameters
1	Change Password	Enter new password or Press the * key for no password
2	Select Greeting to use	Greeting #1 (default) Greeting #2
3	Record Greeting #1	Record greeting and press # when finished
4	Record Greeting #2	Record greeting and press # when finished
5	Record your name	Record your name and press # when finished (this is the recording played by a directory in the Auto Attendant)
6	Date And Time Stamp On/Off	On Off (default)
7	Set whether callers can leave a message	1 Allow messages to be left (default) 2 Disallow messages to be left
8	Modify Distribution Group Settings	Select group number to modify: 1 Add user to group 2 Hear list of users in group 3 Remove all users from group
9	Select Transfer Type	1 Defined by Auto Attendant (default) 2 Blind Transfer 3 Screen Caller
10	Set Busy Options, *Intercept MSG*	1 Sends caller directly to mailbox 2 Play custom message 3 Play default intercept message (default)
11	Set options for remote Call Forwarding	1 Immediate 2 If no answer 3 Disabled (default)
12	Select the phone number for Remote Call Forwarding	Select a phone number to be used. Enter a Number from 1 - 4. This number will be from the Remote Call Forwarding and Remote Call Notification phone number list. 1 (default)
13	Set Options for Remote Call Notification	1 Notify phone of new messages 2 Disable (default)
14	Select the Phone Number for Remote Call Notification	Select a phone number to be used. Enter a Number from 1 - 4. This number will be from the Remote Call Forwarding and Remote Call Notification phone number list. 1 (default)
15	Set Options for Digital Pager Notification	1 Notify page of new messages 2 Disable (default)
16	Enter the Remote Call Forwarding and Remote Call Notification Phone Number	Remote Call Forwarding and Remote Call Notification List 1 - 4 Four separate telephone numbers may be entered to be used with Remote Call Forwarding and Remote Call Notification. Only one number may be used at a time.
17	Enter Digital Pager Phone Number	To Send a 2 Second Pause Press *1 To Send a * Press ** To Send a number Press * followed by the number
18	Specify the number of times the system will attempt to call a Digital Pager or Remote Call Notification Phone Number	1 - 9 attempts or select 0 for infinite number of attempts 5 attempts (default)
19	Specify the number of minutes between retry attempts for Digital Pager or Remote Call Notification	1 - 99 minutes 15 minutes (default)

9.4.11 Recording mailbox greetings

Prerecorded .wav files may be incorporated into the mailbox parameters or recordings may be made using the Plexus Administrator audio utility. In order to use the integrated audio utility the PC must have the following components properly configured for use within Windows:

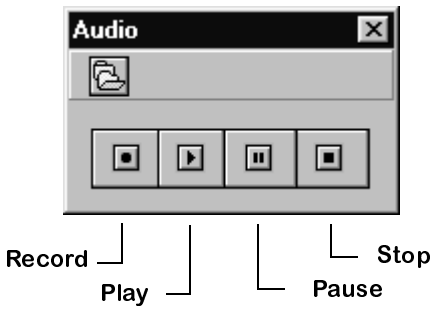
- sound card
- speakers
- microphone

To make a recording using the integrated audio utility, proceed as follows:

1 Click on one of the following buttons on the Mail-Box Editor dialog window to launch the audio utility:

- Name
- Greeting 1
- Greeting 2

9. The audio utility will launch, as seen below.



2 Click on the Record button.

3 Speak into the microphone.

4 Click again on Record (or click on Stop) when complete.

5 The sound recording will be automatically saved and indexed as part of the mailbox configuration file.

Note
The integrated audio utility produces **PCM 8KHz 16 Bit Mono** .wav files. Prerecorded .wav files must be converted to the above format in order to be utilized in the voice mail system.

To incorporate a previously recorded .wav file, proceed as follows:

1. Click on in the Audio dialog window.
2. Select the desired .wav file.
3. Click on OK.

9.4.12 Saving mailboxes

To save a mailbox configuration developed using the Plexus Administrator **VPS Mail Editor**, select **Save As** from the **File** menu and name the file.

Note
Plexus Administrator mailbox configuration files have an extension of **.xdb**. E.g., The Configuration Wizard creates the mailbox configuration file **wiz-mail.xdb**

The mailbox configuration file need only be given a name the first time it is saved. Subsequent saves may be accomplished simply by selecting **Save** under the **File** menu.

9.4.13 Formatting mailboxes

Before a completed mailbox configuration can be uploaded to the Plexus system, the mailbox configuration must be converted to .vps file format. This is accomplished via the **Format** menu option.

The **Format** menu option may be used to format either an entire mailbox configuration or selected mailboxes only. This feature enables quick and easy mailbox additions and modifications.

To format either an entire mailbox configuration or selected mailboxes, proceed as follows:

1. Save the mailbox configuration or open an existing mailbox configuration file.
2. If only selected mailboxes will be formatted, select the desired mailboxes by clicking once on the corresponding Mail-Box ID.

3 The selected mailboxes will become highlighted.

Note
Formatting a single mailbox or selected mailboxes will only affect those formatted mailboxes. Other mailboxes already uploaded to the Plexus system will not be affected.

Note
<Ctrl> and **<Shift>** may be used to select multiple mailboxes. Click on the first mailbox so that it becomes highlighted. Hold down **<Ctrl>** and select additional mailboxes. Hold down **<Shift>** and use **<Ç >** and **<È >** to select a consecutive list of mailboxes.

4 From the **File** menu, select **Format**.

5 The following pull-down menu will be displayed.

6 Point to **All Items** or **Selected Item(s) Only** and click once.

7 Enter a name for the new .vps file.

8 Click on **OK**.

Note
The Node Editor and the Mail-Box editor both format files using .vps file format. If you format your auto attendant structure and your mailbox structure with the same name, one file will overwrite the other. Be sure to give unique names to each .vps file.

9.4.14 Uploading

1 Launch the Plexus Administrator Windows application.

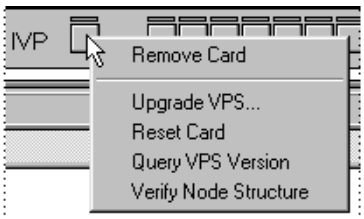
2 Open the system's configuration file (**.zdb** file)

3 From the **Link** menu, select **Open**.

4 The link indicator at the bottom of the screen should indicate **Link: Opened** and the LED image should appear green. (See Software Configuration - "Setting up a link" for more information).

5 Right click on the IVP card.

6 From the pop-up menu, select “Upgrade VPS”.



7 Locate the voice mailbox .vps file.

8 Click on **OK**.

9 It may take several minutes for the upload to complete. A status indicator at the bottom of the screen will display the progress of the upload. When completed, a window will popup with **VPS Upgrade Complete.** Click on **OK**.

9.5 System Parameters

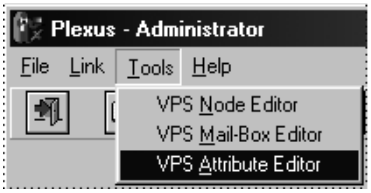
9.5.1 Description

System Parameters are IVP settings that affect the system. They include the Voice Mail System Manager password, the Silence Detect Threshold among others. These two settings can be set by the VPS Attribute Editor. The others must be modified through the Voice Mail System Manager (see “Change system parameters” - section 9.6.7).

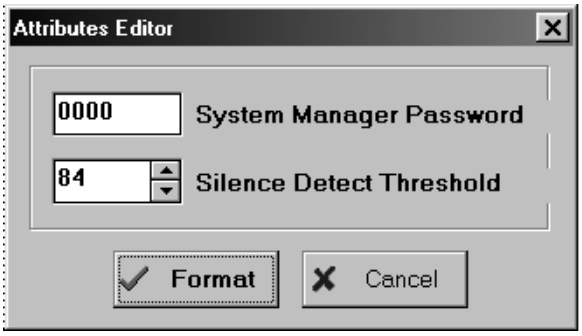
9.5.2 Modifying the system parameters with the VPS Attribute Editor

To use the VPS Attribute Editor:

1. Launch Plexus Administrator. Refer to the **Software Configuration** section of the Plexus Product Manual for instructions on launching Plexus Administrator.
2. From the **Tools** menu, select **VPS Attribute Editor**.



3. The Attribute Editor dialog window will be displayed with default settings.



4. After modifying the settings, press **Format**. A file named **system.vps** will be automatically created in the Plexus\saved\ folder.

9.5.2.1 System Manager Passwords

Modifying the system manager password will only change the password to mailbox 001.

Note
Only the Voice Mail System Manager logged into Voice Mail System Manager Mailbox #001 can change the passwords for all five of the System Manager Mailboxes.

The default password for all System Manager Mailboxes is 0000.

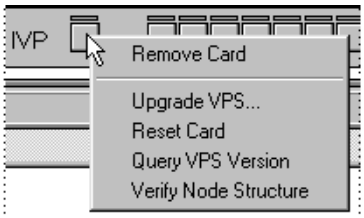
Be sure to keep a record of any changes made to the System Manager Mailbox passwords. Without the proper password, access to the mailboxes is prohibited.

9.5.2.2 Silence Threshold

While recording a message, the voice mail system will detect when the caller has been silent for a certain period of time. This parameter sets the voice level which is presumed to be silence. A lower number means that the incoming caller must be quieter for silence to be detected. A higher number may be used when there is a greater amount of background noise on the incoming lines and silence is never detected. This parameter will accept values between 1 to 999. The default value is 84.

9.5.3 Uploading System Parameter changes

1. Launch the Plexus Administrator Windows application.
2. Open the system's configuration file (**.zdb** file)
3. From the **Link** menu, select **Open**.
4. The link indicator at the bottom of the screen should indicate **Link: Opened** and the LED image should appear green. (See Software Configuration - "Setting up a link" for more information).
5. Right click on the IVP card.
6. From the pop-up menu, select "Upgrade VPS".



7 Locate the **system.vps** file. It will be in the saved folder in the folder which the Plexus Administrator was installed in.

8 Click on **OK**.

A status indicator at the bottom of the screen will display the progress of the upload. When completed, a window will popup with **VPS Upgrade Complete.** Click on **OK**.

9.6 Voice Mail System Manager

9.6.1 Description

It is likely that a dealer will initially configure each Plexus voice mail system. However, trained customer personnel can handle much of the day-to-day maintenance issues. Such individuals serve as Voice Mail System Managers and perform several tasks including:

"Make changes to the auto attendant system" - section 9.6.5

"Make changes to the voice mailboxes" - section 9.6.6

"Change system parameters" - section 9.6.7

"Hear the amount of space left on the hard drive" - section 9.6.8

"Hear the version numbers of the voice mail system" - section 9.6.9

9.6.2 Authorization levels

Before an individual can configure any part of the voice mail system via a DTMF-capable telephone, they must log in to one of the System Manager Mailboxes. There are five System Manager Mailboxes, each with its own unique password and privileges. The different mailboxes allow the Voice Mail System Manager to give restricted administrative privileges to other personnel. The System Manager Mailboxes and the associated privileges are as follows:

Mailbox #	Administrative Privileges
005	Configuration of auto-attendant nodes, including the addition, deletion, and modification of nodes.
004	All the privileges of 005, plus add and modify mailboxes.
003	All the privileges of 004, plus delete mailboxes.
002	All the privileges of 003, plus modify user-side mailbox parameters without requiring entry of a password, listen to messages left in any mailbox, and hear the summary of new messages at all mailboxes.
001	All the privileges of 002, plus change passwords for all system manager mailboxes, delete all auto-attendant nodes, delete all mailboxes, and rerecord system messages.

Note
The default password for all System Manager Mailboxes is 0000. If trained customer personnel will be given access to these mailboxes, it is strongly suggested that the personnel receive only the mailbox # access needed to complete the required tasks and that all System manager mailboxes receive new passwords. See “System Parameters” - section 9.5.

9.6.3 Accessing the system

Access to the Voice Mail System Manager can be done locally (on a phone attached to the system) or remotely (calling into the system from an external location).

9.6.3.1 Accessing the Voice Mail System Manager locally:

A. Accessing From a Plexus Key Telephone

To access the voicemail system from a Plexus Key Telephone, proceed as follows.

1. From a Key Telephone, press **MSG**.
2. The voice mail system will prompt you to log into a mailbox or enter your mailbox password.
3. Press ‘*****’ followed by the appropriate mailbox (001-005).
4. When prompted for a password, enter the appropriate password.
5. After completing these steps, the extension (i.e., user) is logged into a System Manager Mailbox.

B. Accessing From an Ordinary Telephone

Alternately, you may access the voicemail system from an ordinary telephone.

1. At system dial tone, enter **<F52>**.
2. The voice mail system will prompt you to log into a mailbox or enter your mailbox password.
3. Press ‘*****’ followed by the appropriate mailbox (001-005).
4. When prompted for a password, enter the appropriate password.
5. After completing these steps, the extension (i.e., user) is logged into a System Manager Mailbox.

9.6.3.2 Accessing the Voice Mail System Manager remotely:

1. Dial into the system as a regular caller.
2. If a user answers, request to be connected to an Auto Attendant or to be transferred into somebody's mailbox.
3. When the Auto Attendant answers or you receive a mailbox greeting, press '*' followed by the appropriate mailbox (001-005).
4. When prompted for a password, enter the appropriate password.
5. After completing these steps the caller is logged into a System Manager Mailbox.

9.6.4 System Manager Main Menu

Once you have logged into a System Manager Mailbox, you will receive the System Manager Main Menu. Complete access to this menu and sub menus is only available when logged into System Manager Mailbox #001. If you try to access a menu choice not available in your mailbox privileges, a message will be played advising that this access is not permitted.

1	"Make changes to the auto attendant system" - section 9.6.5.
2	"Make changes to the voice mailboxes" - section 9.6.6.
3	"Change system parameters" - section 9.6.7.
4	"Hear the amount of space left on the hard drive" - section 9.6.8.
5	"Hear the version numbers of the voice mail system" - section 9.6.9.
0	To hear this menu again.
*	To enter another mailbox.

9.6.5 Make changes to the auto attendant system

Upon reaching the System Manager Main Menu, select '**1**' to hear the Auto Attendant Main Menu

9.6.5.1 Auto-Attendant Main Menu

To create a node.

To listen to a description of an existing node.

To delete a node.

To delete all nodes.

To make a copy of an existing node.

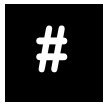
To set the node starting points.

To listen to information about how to program the auto attendant system.

To enter the auto attendant system as an incoming caller.

To record a node message (hidden option)

To enter another mailbox.



To go back to the previous menu.

9.6.5.2 Create a Node

This feature is accessible to All System Manager Mailboxes.

To create a node, proceed as follows:

1. Select **'1'** from the Auto-Attendant Main Menu.
2. Enter a node number. If the node number is less than eight digits, press the '#'key to end the entry.
3. Press the digit corresponding to the desired type of node.

1	To create a Menu Node.
2	To create a Dial Node.
3	To create a Play Message Node.
4	To create a Get Extension Node.
5	To create a Mailbox-Access Node.
6	To create an Interrogate Node.
7	To create a Directory Node.

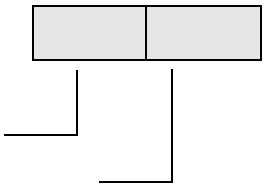
Note
For information on setting up the parameters for each node type created, see "Node Templates" - section 9.6.5.10

9.6.5.3 Listen to a Description of an Existing Node

This feature is accessible from all System Manager Mailboxes.

To hear a description of any existing node, proceed as follows:

1. Select **'2'** from the Auto-Attendant Main Menu.
2. Enter a node number. If the node number is less than eight digits, press the '#' key to end the entry.
3. To hear a listing of all nodes press '*****'.
4. While reviewing the list of all nodes, press '1' to skip to the next node in the list.
5. To hear a listing of all nodes beginning at a certain starting point, enter the two digit starting point followed by '*****'.



6. To exit this menu, press '#'.

9.6.5.4 Delete a Node

This feature is accessible from all System Manager Mailboxes.

To delete a node, proceed as follows:

1. Select **'3'** from the Auto-Attendant Main Menu.
2. Enter a node number. If the node number is less than eight digits, press the '#' key to end the entry.
3. To confirm deletion of this node, press '1'. To return to the Auto-Attendant Main Menu, press any other key.

Delete All Nodes

This feature is accessible from **System Manager Mailbox 001.**

To delete all nodes, proceed as follows:

1. Select **'4'** from the Auto-Attendant Main Menu.
2. The voice mail system will announce "This will cause all nodes in the auto-attendant system to be deleted."
3. If you want to proceed with this operation, press '1'. To return to the main menu, press any other key.

9.6.5.5 Make a Copy of an Existing Node

This feature is accessible from all System Manager Mailboxes.

To make a copy of an existing node, proceed as follows:

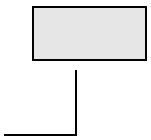
1. Select **'5'** from the Auto-Attendant Main Menu.
2. Enter the node number of the existing node which you would like to copy.
3. If the node number is less than eight digits, press the '#' key to end the entry. **Or**, to return to the main menu, press the '*' key.
4. Enter the node number of the node you want to create.
5. If the node number is less than eight digits, press the '#' key to end the entry. **Or**, to return to the main menu, press the '*' key.

9.6.5.6 Set the Node Starting Points (Route Point Starting Point)

This feature is accessible from all System Manager Mailboxes.

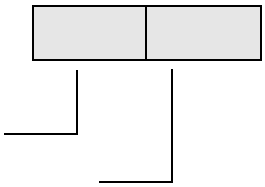
To set the node starting points (i.e., Voice Mail Starting Point), proceed as follows. To exit at any time, press the '#' key.

1. Select **'6'** from the Auto-Attendant Main Menu.
2. Enter the PBX designated starting point (i.e., Route Point) for which you wish to set the Voice Mail Starting Point.



3 The voice mail system will announce "The current setting for PBX starting point ___ (i.e., Route point) is ___ (i.e., Voice Mail Starting Point). This means that an incoming call with this PBX starting point will start at node ___."

4 Enter the new Voice Mail Starting Point to be used with this PBX starting point (route point). Or, to cancel this command, press the '#' key.



9.6.5.7 Listen to Information About how to Program the Auto Attendant System

This feature is accessible from all System Manager Mailboxes.

To access this feature, proceed as follows:

1 Select **'7'** from the Auto-Attendant Main Menu.

2 Listen to prompts for general information about programming the auto attendant system.

9.6.5.8 Enter the Auto-Attendant as an Incoming Caller

This feature is accessible from all System Manager Mailboxes.

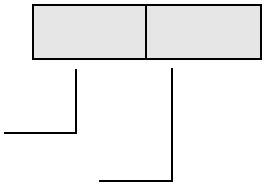
To enter the auto-attendant system as an incoming caller, proceed as follows:

1 Select **'8'** from the Auto-Attendant Main Menu.

2 The voice mail system will announce "This option allows you to listen to a programmed auto-attendant sequence before it is enabled for public use."

3 Enter the two digit starting node.

4 Once the two digit starting node is entered, the auto-attendant system is launched as it would be for an incoming caller. All keys apply as they would for the defined nodes. Or, to return to the Auto-Attendant Main Menu, press the '#' key.



9.6.5.9 To record a node message (hidden option)

This feature is accessible from all System Manager Mailboxes.

To access this feature, proceed as follows:

1 Select **'9'** from the Auto-Attendant Main Menu.

2 Enter the node number for which you wish to make a recording. Only Menu Nodes, Play Message Nodes, Directory Nodes (with custom message), and Interrogate Nodes have message recordings. If the node number is less than eight digits, press the '#' key to end the entry.

3 Record the message. Speak clearly in a normal to soft voice. A loud recording or a recording with shrill, high notes can interfere with the Plexus system's DTMF recognition.

4 Press '#' when finished recording.

5 To rerecord the message, repeat these steps.

9.6.5.10 Node Templates

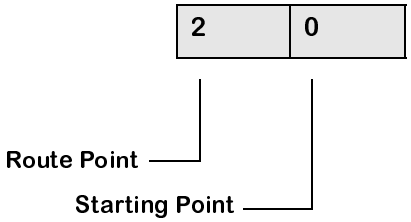
A. Menu Node Template

This section provides a template for creating and configuring the Menu Node from a Plexus key telephone. Each of the steps is described more fully below.

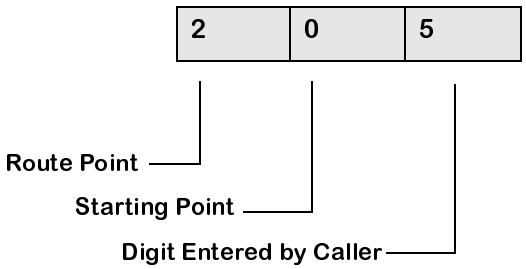
1. Enter the Node Number.
2. Select the action to be taken when the incoming caller presses the '*' key.*
3. Define the action to be taken when the incoming caller presses the '#' key.
4. Record your message after the following tone. Press the '#' key when finished recording.

Node Number

The voice mail system will route a call from a Menu Node to a subsequent node based on the digit entered by the caller. For example, the node number for a Menu Node with a Route Point of '2' and a Voice Mail Starting Point of '0' is '20'.



If the caller presses '5' after listening to the menu greeting, the call will be routed to node '205'. If node '205' exists, the voice mail system will perform the function associated with this node.



Note
If node '205' does not exist a short error message (i.e., "The digit you dialed is not a valid choice") is played, and the Menu Node is restarted.

Select Action for * Key

For the Menu Node, you can select the action to be taken when the incoming caller presses the '*' key.

Note
Options 4, 5, and 6 are only available if the voicemail has been equipped with the Multiple Languages feature.

1 To allow the incoming caller to press the '*' key followed by a mailbox number to log into that mailbox.

2 To take no action and repeat the menu message.

3 To restart the auto-attendant at the initial node.

4 To allow the incoming caller to press the * key followed by a mailbox number to log into that mailbox. This also enables the caller to select from any available language nodes.

5 To take no action and repeat the menu message. This also enables the caller to select from any available language nodes.

6 To restart the auto-attendant at the initial node. This also enables the caller to select from any available language nodes.

Select Action for # Key

You can define the action to be taken when the incoming caller presses the '#' key.

1 To allow the incoming caller to press the '#' key followed by an extension number to leave a message for that extension.

2	To take no action and repeat the menu message.
3	To restart the auto-attendant at the initial node.

Record Message

Record your message after the following tone. Press the '#' key when finished recording.

After recording your message, press one of the following keys:

1	To confirm the recording.
2	To listen to the recording.
3	To re-record the message.
Any Key	Press any other key to return to the main menu.

B. Dial Node Template

This section provides a template for creating and configuring the Dial Node from a Plexus key telephone. Each of the steps is described more fully below.

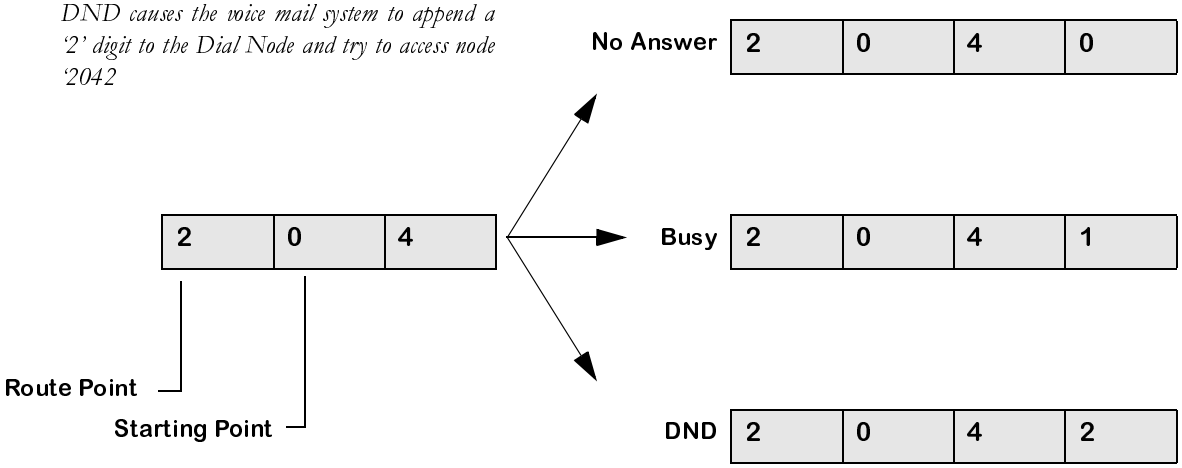
1. Enter the Node Number.
2. For the Dial Node, you can specify the extension number or the PBX group number to which you want the call to be transferred.
3. Define the transfer type.
4. If transfer type 4 is selected, enter the node number that will be accessed after this node. If the node number is less than eight digits, press the '#' key after entry.

Node Number

When a Dial Node attempts a supervised transfer to an unavailable user, or attempts a blind transfer with voicemail as the coverage, the voice mail system will route the call to a subsequent node based on the unavailable state (no answer, busy, DND) of the user.

Example

- *If Dial Node, '204', receives no answer when attempting to transfer a call to a user, the voice mail system will try to access node '2040'.*
- *An attempt to transfer a call to a busy user, will cause the voice mail system to append a '1' digit to the Dial Node and try to access node '2041'.*
- *Finally, an attempt to transfer a call to a user in DND causes the voice mail system to append a '2' digit to the Dial Node and try to access node '2042*



Specify Extension Number of PBX Group

For the Dial Node, you can specify the extension number or the PBX group number to which you want the call to be transferred.

User ID or User Group ID.

If you want to use an extension number which will be entered by the incoming caller as part of a previous Get Extension or Directory Node.

Transfer Type

1 To dial the extension in the blind transfer mode.

2 To ask the incoming caller their name and announce this to the called extension (i.e., user).

4 To end this node and start the next node when the called extension (i.e., user) answers. (For future release.)

C. Play Message Node Template

This section provides a template for creating and configuring the Play Message Node from a Plexus key telephone. Each of the steps is described more fully below.

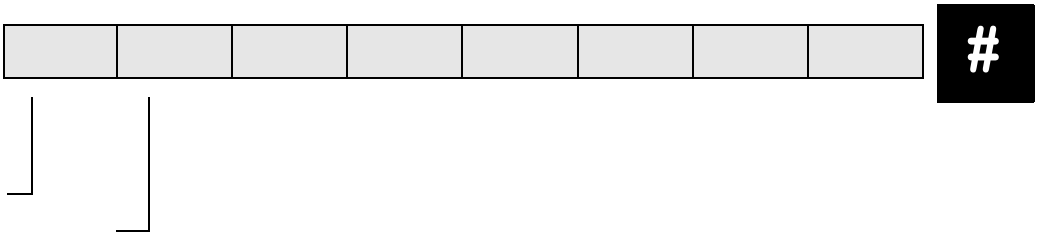
1. Enter the Node Number.
2. Enter the node number that will be accessed after this node. If the node number is less than eight digits, press the '#' key to end the entry.
3. Set the Play Message Node parameters.
4. Record your message.
5. Enter the number of seconds for this operation.
6. Complete the message recording.

Node Number

Enter the Node Number

Enter Subsequent Node Number

Enter the node number that will be accessed after this node. If the node number is less than eight digits, press the '#' key to end the entry.



If you want the auto-attendant system to end the call after accessing this Play Message Node.

1 **#** If you want the auto-attendant system to start from its initial node after accessing this Play Message Node.

Play Message Node Parameters

1 To record your own message for this node.

2 To cause the auto-attendant to say the digits of the extension entered in a previous Get Extension or Directory Node.

3 To not say any message and advance immediately to the next node.

4 To play the name of the last incoming caller.
(This feature will be fully implemented in a future release.)

5 To give silence for a specified amount of time.

6 To play the Music-On-Hold for a specified amount of time.

7 To cause the auto-attendant to say the name of the mailbox owner of the extension entered in a previous Get Extension or Directory Node.

Record Message

1. If 1 is selected above, record your message after the following tone.
2. If 5 is selected above, enter the number of seconds for this operation. You may enter any number from '1' to '999', followed by the '#' key. To exit, press the '*' key.
3. Press the '#' key when finished recording.
4. After recording your message, press one of the following options, or press any key to go to the main menu.

1 To confirm the recording.

2 To listen to the recording

3 To re-record the greeting

D. Get Extension Node Template

This section provides a template for creating and configuring the Get Extension Node from a Plexus key telephone. Each of the steps is described more fully below.

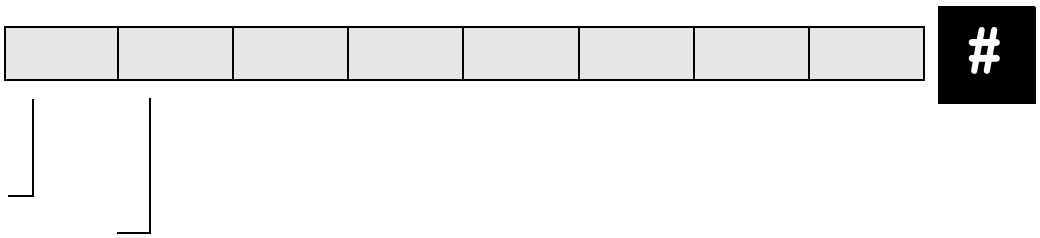
1. Enter the Node Number.
2. Enter the node number that will be accessed after this node. If the node number is less than eight digits, press the '#' key to end the entry.
3. Get Extension Node parameters. Enter the number of digits which are to be entered by the incoming caller as part of this node.

Node Number

Enter the node number

Enter Subsequent Node Number

Enter the node number that will be accessed after this node. If the node number is less than eight digits, press the '#' key to end the entry.



If you want the auto-attendant system to end the call after accessing this Get Extension Node.

If you want the auto-attendant system to start from its initial node after accessing this Get Extension Node.

Get Extension Node Parameters

Enter the number of digits which are to be entered by the incoming caller as part of this node.

E. Mailbox Access Node Template

This section provides a template for creating and configuring the Mailbox Access Node from a Plexus key telephone. Each of the steps is described more fully below.

1. Enter the Node Number.
2. Enter the node number that will be accessed after this node. If the node number is less than eight digits, press the '#' key to end the entry.

3 For the Mailbox Access Node, enter the mailbox number which is to be accessed by this node.

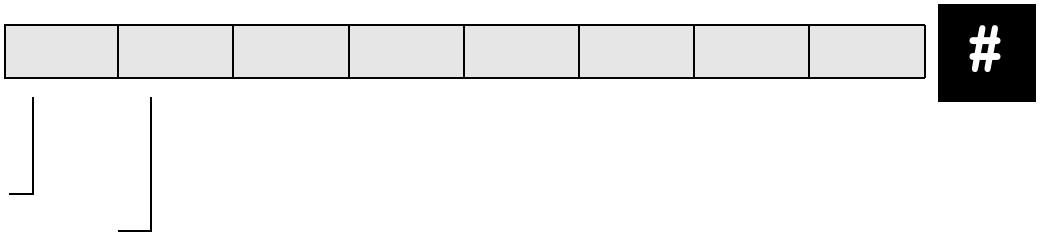
4 Allow a message to be left, or simply play a greeting.

Node Number

Enter the node number

Enter Subsequent Node Number

Enter the node number that will be accessed after this node. If the node number is less than eight digits, press the '#' key to end the entry.

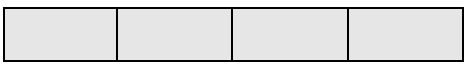


If you want the auto-attendant system to end the call after accessing this Mailbox Access Node.

1 **#** If you want the auto-attendant system to start from its initial node after accessing this Mailbox Access Node.

Enter Mailbox Number

For the Mailbox-Access Node, enter the mailbox number which is to be accessed by this node.



If you want the auto-attendant system to use the mailbox specified by the incoming caller in a previous Get Extension or Directory Node.

Allow Message or Play Greeting

1	To allow the incoming caller to leave a message in the mailbox.
Any Key	To just play the greeting and not allow a message to be recorded by the incoming caller.

F. Interrogate Node Template

This section provides a template for creating and configuring the Interrogate Node from a Plexus key telephone. Each of the steps is described more fully below.

1. Enter the Node Number.
2. Enter the mailbox number where the Interrogate Node will leave the message
3. Enter the node number that will be accessed after this node. If the node number is less than eight digits, press the '#' key to end the entry
4. Record Message.

Node Number

Enter the node number.

Interrogate Node Parameters

1	If this is the last Interrogate Node in the series.
Any Key	If there are more Interrogate Nodes which will be presented to the caller after this one.

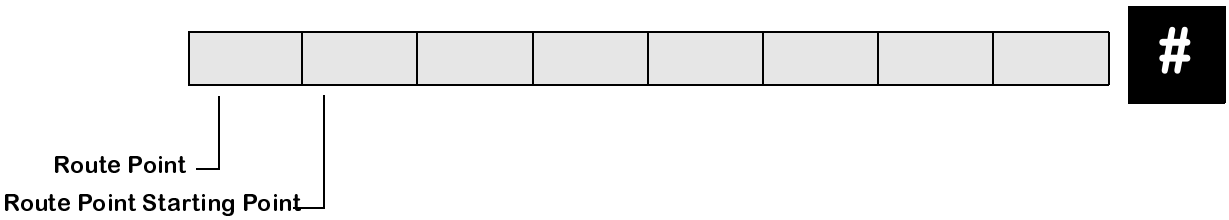
Select Mailbox for Message

If 1 is selected above, enter the mailbox number where the Interrogate Node will leave the message.

If you want the system to use the mailbox entered by the incoming caller in a previous Get Extension or Directory Node.

Enter Subsequent Node Number

Enter the node number that will be accessed after this node. If the node number is less than eight digits, press the '#' key to end the entry.



If you want the auto-attendant system to end the call after accessing this [Interrogate] Node.

1 **#** If you want the auto-attendant system to start from its initial node after accessing this [Interrogate] Node.

Record Message

1. Record your message after the following tone.
2. Press the '#' key when finished recording.
3. When you are finished recording, select from the following options, or press any key to return to the main menu.

1	To confirm the recording.
2	To listen to the recording.
3	To re-record the greeting.

G. Directory Node Template

This section provides a template for creating and configuring the Directory Node from a Plexus key telephone. Each of the steps is described more fully below.

1. Enter the Node Number.
2. Set Directory Node parameters.
3. If you do not want to use the default message, record a new prompt for your Directory node.

Node Number

Enter the node number.

Directory Node Parameters

1	To cause the Directory Node to say each mailbox's name and number.
2	To cause the Directory Node to only say each mailbox's name.

Select Message Prompt for Incoming Calls

The Directory Node will prompt the incoming caller to enter the first three letters of the desired party's last name and the Directory Node will say only those mailboxes.

1 To prompt the incoming caller using the default system message.

2 To prompt the incoming caller using a message that you record.

Record Prompt for Incoming Calls

1. If 2 is selected above, record your message after the following tone.
2. Press the '#' key when finished recording.
3. When you are finished recording, select from the following options, or press any other key to return to the main menu.

1 To confirm the recording.

2 To listen to the recording.

3 To re-record the greeting.

Upon reaching the System Manager Main Menu, select **'2'** to hear the System Manager Mailbox Menu.

9.6.6.1 System Manager Mailbox Menu

7R FUHDWH D QHZ PDLOER[

7R PRGLI\ DQ H[LVWLQJ PDLOER[

7R GHOHWH D PDLOER[

7R GHOHWH DOO PDLOER[HV

7R KHDU D OLVWLQJ RI DOO PDLOER[HV

9.6.6.2 Create a New Mailbox

This feature may be performed by System Managers.

To create a new mailbox, proceed as follows:

1. Select **'1'** from the System Manager Mailbox Menu.
2. To create a single mailbox, enter the mailbox number. This number may be 2-4 digits depending on the configuration.
3. To create a range of mailboxes, press '#'.
4. To return to the main menu, press '*****'.
5. If '#' was selected above, enter the starting mailbox number. Or, to return to the main menu, press '#'.
6. Enter the ending mailbox number.

7 If there are mailboxes which already exist in this range, they will not be modified.

Note
For information on setting up the parameters for each mailbox created, see "Mailbox Templates" - section 9.6.6.7. If a range is created, each mailbox must be configured by selecting option '2', modify an existing mailbox, under the System Manager Mailbox Menu.

0RGLI\ DQ ([LVWLQJ 0DLOER[

This feature may be accessed by System Managers.

To modify an existing mailbox, proceed as follows:

1 Select **'2'** from the System Manager Mailbox Menu.

2 Enter the mailbox number.

3 At this point the voice mail system will proceed with the configuration of the mailbox [Refer to the "Mailbox Templates" - section 9.6.6.7].

'HOHWH D 0D LOER[

This feature may be accessed by System Managers.

To delete a mailbox, proceed as follows:

1 Select **'3'** from the System Manager Mailbox Menu.

2 Enter the mailbox number.

3 To delete this mailbox, press '1'. **Or**, to retain this mailbox, press any other key.

'HOHWH $OO 0DLOER[HV

This feature may be accessed by System Managers.

To delete all mailboxes, proceed as follows:

1 Select **'4'** from the System Manager Mailbox Menu.

2 The voice mail system will announce "This will cause all mailboxes to be deleted and all mailbox messages will be lost."

3 If you want to proceed with this operation, press '1'. **Or,** to return to the main menu, press any other key.

+HDU D /LVWLQJ RI $OO 0DLOER[HV

To hear a listing of all mailboxes, proceed as follows:

1 Select **'5'** from the System Manager Mailbox Menu.

2 The voice mail system will announce the list of mailboxes as follows:
"Mailbox number _ _ _ _ + name of owner (if recorded)"
"Mailbox number _ _ _ _ + name of owner (if recorded)"
etc.

3 While listening to the listing, press '1' to skip to the next mailbox in the list. **OR**, to return to the main menu, press any other key.

0DLOER[7HPSODWHV

The templates on the following pages were developed to assist with configuring mailboxes. Utilize the templates to identify the parameters that must be set up for each mailbox.

0DLOER[7\SH 7HPSODWH

1	7R VHOHFW ZKHWKHU WKLV LV WR EH DQ RUGLQDU\ PDLOER[RU D *URXS 'HO ER[
8	7R VNLS WR WKH QH[W SDUDPHWHU
9	7R JR WR WKH SUHYLRXV SDUDPHWHU
#	7R UHWXUQ WR WKH PDLQ PHQX
0	7R KHDU WKLV PHQX DJDLQ

After selecting option 1, you will be able to select whether this is to be an ordinary mailbox or a Group Delivery mailbox.

1	7R FRQILJXUH WKLV DV DQ RUGLQDU\ PDLOER[
2	7R FRQILJXUH WKLV DV D *URXS %URDGFDVW PDLOER[ZKLFK ZLOO VH HYHU\ QHZ PHVVDJH WR DOO PHPEHUV RI LWV JURXSV
3	7R FRQILJXUH WKLV DV D *URXS 'HOLYHU\ PDLOER[ZKLFK ZLOO VHQG D PHVVDJH WR WKH GLIIHUHQW JURXSV LQ D GHOD\HG RUGHU XQWLO F PHVVDJH
4	7R FRQILJXUH WKLV DV D VLPSOLILHG PDLOER[
	7KH FXUUHQW VHWWLQJ LV GHIDXOW BBBBB
8	7R VNLS WR WKH QH[W SDUDPHWHU
9	7R JR WR WKH SUHYLRXV SDUDPHWHU
#	7R UHWXUQ WR WKH PDLQ PHQX
0	7R KHDU WKLV PHQX DJDLQ

$VVRFLDWHG ([WHQVLRQ 8VHU 7HPSODWH

1	7R VHOHFW ZKHWKHU DQ H[WHQVLRQ L H XVHU LV DVVRFLDWHG ZLWK WK
8	7R VNLS WR WKH QH[W SDUDPHWHU
9	7R JR WR WKH SUHYLRXV SDUDPHWHU
#	7R UHWXUQ WR WKH PDLQ PHQX
0	7R KHDU WKLV PHQX DJDLQ

Associating an extension with the mailbox allows the Dial node to request a transfer to that extension.

1 ,I WKHUH LV WR EH DQ H[WHQVLRQ DVVRFLDWHG ZLWK WKLV PDLOER

2 ,I QRW

7KH FXUUHQW VHWWLQJ LV BBBBB GHIDXOW

8 7R VNLS WR WKH QH[W SDUDPHWHU

9 7R JR WR WKH SUHYLRXV SDUDPHWHU

7R UHWXUQ WR WKH PDLQ PHQX

0 7R KHDU WKLV PHQX DJDLQ

3ULYDWH 0DLOER[7HPSODWH

1	7R VSHFLI\ ZKHWKHU WKLV LV D SULYDWH PDLOER[
8	7R VNLS WR WKH QH[W SDUDPHWHU
9	7R JR WR WKH SUHYLRXV SDUDPHWHU
#	7R UHWXUQ WR WKH PDLQ PHQX
0	7R KHDU WKLV PHQX DJDLQ

A mailbox specified as private will not be announced to the incoming caller during a Directory node.

1	7R FRQILJXUH WKLV PDLOER[DV SULYDWH
2	7R DOORZ WKLV PDLOER[WR EH SXEOLFO\ YLVLEOH
	7KH FXUUHQW VHWWLQJ LV GHIDXOW BBBBB
8	7R VNLS WR WKH QH[W SDUDPHWHU
9	7R JR WR WKH SUHYLRXV SDUDPHWHU
#	7R UHWXUQ WR WKH PDLQ PHQX
0	7R KHDU WKLV PHQX DJDLQ

0HVVDJH 6XPPDU\ 7HPSODWH

1	7R VSHFLI\ ZKHWKHU WKLV PDLOER[FDQ KHDU D VXPPDU\ RI PHVVDJHV LQ D ER[HV
8	7R VNLS WR WKH QH[W SDUDPHWHU
9	7R JR WR WKH SUHYLRXV SDUDPHWHU
#	7R UHWXUQ WR WKH PDLQ PHQX
0	7R KHDU WKLV PHQX DJDLQ

After selecting option 1 above, you will be able to select from the following options:

1	7R DOORZ WKLV PDLOER[WR KHDU D VXPPDU\ RI QHZ PHVVDJHV LQ DC
2	7R 127 DOORZ WKLV PDLOER[WR KDYH WKLV FDSDELOLW\
	7KH FXUUHQW VHWWLQJ LV BBBBB GHIDXOW
8	7R VNLS WR WKH QH[W SDUDPHWHU
9	7R JR WR WKH SUHYLRXV SDUDPHWHU
#	7R UHWXUQ WR WKH PDLQ PHQX
0	7R KHDU WKLV PHQX DJDLQ

3DJHU 1RWLILFDWLRQ 7HPSODWH

1	7R VSHFLI\ ZKHWKHU WKLV PDLOER[LV DOORZHG WR KDYH WKH V\VWHP FDO EHU DIWHU UHFHLYLQJ D QHZ PHVVDJH
8	7R VNLS WR WKH QH[W SDUDPHWHU
9	7R JR WR WKH SUHYLRXV SDUDPHWHU
#	7R UHWXUQ WR WKH PDLQ PHQX
0	7R KHDU WKLV PHQX DJDLQ

After selecting option 1 above, you will be able to select from the following options:

1	7R DOORZ WKLV H[WHQVLRQ WR XVH WKH SDJHU QRWLILFDWLRQ IHD
2	7R 127 DOORZ WKLV H[WHQVLRQ WR XVH WKH SDJHU QRWLILFDWLRQ I VHWWLQJ LV BBBBB GHIDXOW
8	7R VNLS WR WKH QH[W SDUDPHWHU
9	7R JR WR WKH SUHYLRXV SDUDPHWHU
#	7R UHWXUQ WR WKH PDLQ PHQX
0	7R KHDU WKLV PHQX DJDLQ

Remote Telephone Notification Template

1	7R VSHFLI\ ZKHWKHU WKLV PDLOER[LV DOORZHG WR KDYH WKH V\VWHP QRW WHOHSKRQH DQG HU DIWHU UHFHLYLQJ D QHZ PHVVDJH
8	7R VNLS WR WKH QH[W SDUDPHWHU
9	7R JR WR WKH SUHYLRXV SDUDPHWHU
#	7R UHWXUQ WR WKH PDLQ PHQX
0	7R KHDU WKLV PHQX DJDLQ

After selecting option 1 above, you will be able to select from the following options:

1	To allow this extension to use the remote telephone number notification feature.
2	To NOT allow this extension to use the remote telephone number notification feature.
	The current setting is _____ (2).
8	To skip to the next parameter.
9	To go to the previous parameter.
#	To return to the main menu.
0	To hear this menu again.

Mailbox Owner's Name Template

1	To record the name of the mailbox owner.
8	To skip to the next parameter.
9	To go to the previous parameter.
#	To return to the main menu.
0	To hear this menu again.

After selecting option 1 above, you will hear the current recording. You may then select to change the recording, or return to the main menu.

1	To change this recording. If option 1 is selected, you may record your message after the tone. Press the '#' key when finished recording.
Any Key	To return to the main menu.

Mailbox Owner's Directory Listing Template

1	To enter the last three letters of the mailbox owner's last name.
8	To skip to the next parameter.
9	To go to the previous parameter.
#	To return to the main menu.
0	To hear this menu again.

After selecting option 1 above, you will hear the current settings. You may then select to change the settings, or return to the main menu.

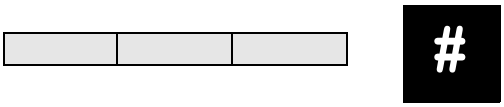
Using the letters associated with the numbers on your telephone, enter 3 digits representing the first 3 digits of the person's last name. You must enter three digits. For example, the last name "Smith" would be entered by dialing 'S', 'M', 'I', or '7', '6', '4'. For last names having fewer than 3 letters, use the zero key at the end of the name to fill the entry. To enter the letter 'Q', use the 7 key. To enter the letter 'Z', use the 9 key.

#	To skip to the next parameter.

Maximum Message Age Template

1	To set the number of days a message can remain in this mailbox until it is automatically deleted.
8	To skip to the next parameter.
9	To go to the previous parameter.
#	To return to the main menu.
0	To hear this menu again.

After selecting option 1 above, you will hear the current setting. To change this setting, you may enter any number from 1 to 365 days followed by the '#' key.



#	If you do want any messages to be automatically deleted.
1	To return to the previous menu without modifying this setting.

Password Template

1	To reset the mailbox password to be the same number as the extension and to erase the two mailbox greetings and use the default system greeting.
8	To skip to the next parameter.
9	To go to the previous parameter.
#	To return to the main menu.
0	To hear this menu again.

Your entry has been saved. There are no more options to be programmed.

9.6.7 Change system parameters

9.6.7.1 Description

Upon reaching the System Manager Main Menu, select **'3'** to hear the System Parameter menu.

Note
The System Parameter menu can only be accessed from mailbox 001.

9.6.7.2 System Parameter Templates

System Manager Passwords Template

1	To change the System Manager Passwords
8	To skip to the next parameter.
9	To go to the previous parameter.
#	To return to the main menu.
0	To hear this menu again.

.

Changing the System Manager Passwords

1 Enter the System Manager level password you would like to change. There are 5 levels, 1 being the highest and 5 being the lowest.

1 or 2 or 3 or 4 or 5

2 Enter your new four-digit password. Or, if you do not want to use a password, press the '#' key.

3 You have entered _ _ _ _.

4 To accept this as your new password, press 1.

5 To return to the main menu, press any key.

Silence Threshold Template

This feature is accessible through the System Manager Mailbox 001.

1	To set the silence detect threshold.
8	To skip to the next parameter.
9	To go to the previous parameter.
#	To return to the main menu.
0	To hear this menu again.

To Change the Silence Threshold

1. Press 1. The system will announce the current setting:
2. The current setting is _____ ($84^{default}$).
3. You may enter any number from '1' to '999', followed by the '#' key. A smaller number makes the system more sensitive and may delay the detection of silence. A larger number makes the system less sensitive and may end the recording prematurely if a caller is speaking too softly.
4. To return to the previous menu without changing this parameter, press the '*' key.

The recommended value for this field is 84.

Maximum Message Age Template

This feature is accessible through the System Manager Mailbox 001.

1	To set the default value for the number of days a message can remain on the system until it is automatically deleted.
8	To skip to the next parameter.
9	To go to the previous parameter.
#	To return to the main menu.
0	To hear this menu again.

To Change the Maximum Message Age

1. After selecting '1' from the menu above, you will hear the current setting. By default, messages are not automatically deleted.
2. The current setting is _____.
3. You may enter any number from '1' to '365' days, followed by the '#' key.
4. Press '#' If you do not want any messages to be automatically deleted.
5. Press '*' to return to the previous menu.

System Messages Template

This feature is accessible through the System Manager Mailbox 001.

1	To record new system messages.
8	To skip to the next parameter.
9	To go to the previous parameter.
#	To return to the main menu.
0	To hear this menu again.

To Play the Current Message

1. Select '1' from the menu above.
2. Enter the three digit message number.
3. The message will be played.
4. To return to the previous menu, press '#'.
5. If you record your own message over the existing message, you will **not** be able to restore the original system message.

1	To record a new message.
2	To listen to the next message.
Any Key	To enter a new message number.

of Digits in Mailbox Number Template

This feature is accessible through the System Manager Mailbox 001.

1	To set the number of digits needed to specify a mailbox.
8	To skip to the next parameter.
9	To go to the previous parameter.
#	To return to the main menu.
0	To hear this menu again.

To Set the Number of Digits

1. After pressing '1' from the menu above, you will hear the following message: "The current setting is x digits per mailbox." ($3^{default}$)
2. Enter the new value for this parameter. You may specify that 2, 3, or 4 digits are required to specify a mailbox number.
3. Then, select from the following options.

8	To skip to the next parameter.
9	To go to the previous parameter.
#	To return to the main menu.
0	To hear this menu again.

9.6.8 Hear the amount of space left on the hard drive

Upon reaching the System Manager Main Menu, select **'4'** to hear the amount of space available for message recording on the IVP hard drive. The following information will be played:

There is approximately _____ hour(s) and _____ minutes of storage space left on the hard disk.

The system reports the number of hours and minutes of recording time remaining on the hard disk. When the amount of space on the hard disk slips below 1 hour of recording time, a warning is presented to all users upon logging into their mailboxes. If the available recording time drops below 15 minutes, the system does not allow any recording to take place and instead plays an error message indicating that the disk is full.

Deleting messages from the hard drive may not immediately free up space. The system recalculates the hard disk status approximately every 6 hours. However, it is possible to force a recalculation of disk space by accessing the Maximum Message Age system parameter menu. When this menu is invoked, the system automatically recalculates the space on the hard disk. Refer to "Modify an Existing Mailbox" - section 9.6.6.3 for more information.

9.6.9 Hear the version numbers of the voice mail system

This feature is accessible through all System Manager Mailboxes. Upon reaching the System Manager Main Menu, select **'5'**.

A string of numbers indicating the voice mail system operating system and various serial numbers will be played. The number immediately following the first "The current setting is" prompt is the current voice mail operating system.

Note
This version number should periodically be compared to the latest available version and upgraded when necessary.

9.7 Setting up Hold Queue Intercepts

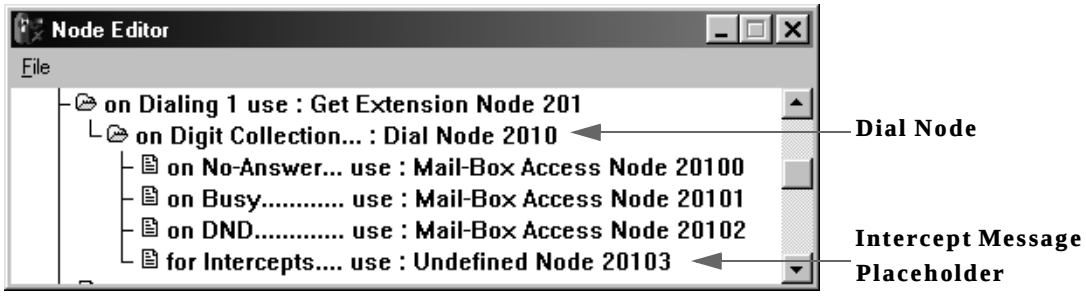
9.7.1 Description

Busy Queuing for a user or user group can be set up such that the queue is intercepted periodically with a default system message, a custom message, or a connection to another user on the system. The default system message provides a point-in-queue indication so that the caller knows how many calls are ahead of them. While custom messages cannot take advantage of the point-in-queue indication, they should provide reassurance to callers, letting them know that the system is aware that they are still holding. If the intercept is directed to another user on the system, that user can update the caller on the availability of the targeted user. Busy Queuing occurs when a call is Camped-On or Parked for the busy user; it is not available to calls placed on hold.

This section covers the configuration of intercepts that provide a default system message or a custom recorded message. Both these options use the IVP to play messages for the holding caller.

9.7.2 Setup Voice ID with Intercept Pilot

For Intercept Messages to work, Intercept Pilots for users and user groups must be defined as a voice ID. This voice ID must have a Intercept Point defined. This Intercept Point ***must*** be a ***Dial Node***. When the system accesses this Intercept Point, it will access the placeholder for Intercept Messages under the Dial Node. This node can be an Undefined Node for system default messages or a Play Message Node or Menu Node for custom messages (see “Custom messages” - section 9.7.5 for information on additional parameters that must be defined). The placeholder is defined by adding a 3 to the defined Dial Node (e.g., dial node 2010 has a Intercept message placeholder of node 20103). For system default messages only, the Intercept Point Dial Node ***must*** be a Dial Node under a Get Extension Node for proper operation.



9.7.2.1 Voice ID configuration

Scroll to the desired Voice ID and make appropriate entries and selections on each of the tabs presented below.

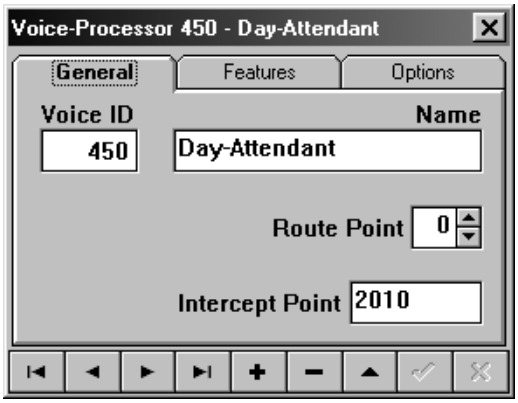


Table 9 - 12 General

Voice ID	
Logical identifier for the voice mail resource. ***Note*:** The ID is automatically assigned based on the Voice ID Sequence entered when the first Voice ID was created.	Numeric 2 to 4 digits, (depending on the dial plan)
Intercept Point	
The main **dial node** over the intercept node.	Numeric

9.7.3 Setup user and user group Intercept Service

To set up intercept service for a user or user group, enable and assign an intercept pilot.

9.7.3.1 User Intercept Pilot

The settings to set up intercept service for a user are found in User: Assignment, Support (see “User: Assignment, Support” - section 6.22.11).

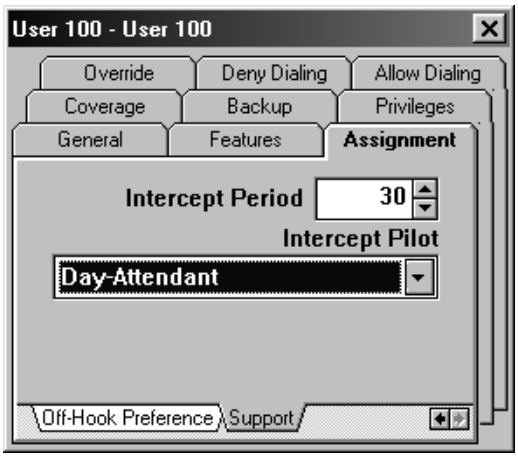


Table 9 - 13 User: Assignment, Support

Intercept Period		
The period of time between connections to the intercept pilot while a caller is waiting in a user’s exclusive hold queue. ***Note:*** See “Intercept Service” - section 8.84.	0-255 seconds	30 seconds*
Intercept Pilot		
Intercept **connect to** for calls in queue for the user.	None * User ID User Group ID Auto Attendant ID Voice ID	

* = default settings

The Intercept Pilot should be set to a voice ID of the IVP card. This voice ID will have an intercept point defined (see “Setup Voice ID with Intercept Pilot” - section 9.7.2).

Users enable Busy Queuing on an individual basis. See “Busy Queuing - User” - section 8.21 for detailed instructions for enabling the busy Busy Queuing with an Intercept Message.

9.7.3.2 User Group Intercept Pilot

The settings to set up intercept service for a user group are found in User Group: Distribution (see “User Group: Distribution” - section 6.23.2).

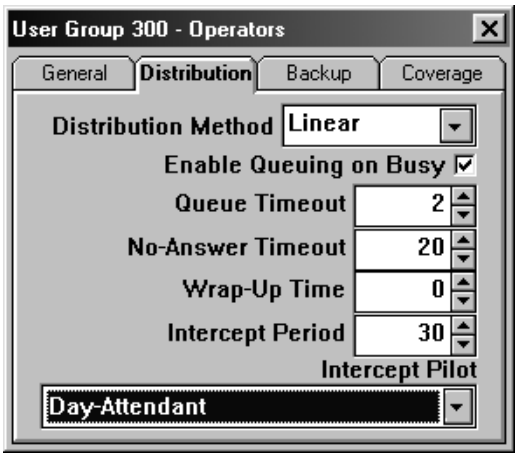


Table 9 - 14 User Group: Distribution

Enable Queuing on Busy		
Whether Busy Queuing is enabled for calls to this group. ***Note:*** See “Busy Queuing - User Group” - section 8.22.	Enable / disable*	
Queue Timeout		
The period of time that a call may remain in queue waiting for an available user before routing to backup and/or coverage. ***Note:*** See “Busy Queuing - User Group” - section 8.22.	0-255 minutes	2 minutes*
Intercept Period		
Refers to the period of time between reassurance messages while a caller is waiting in the busy queue.	0-255 seconds	30 seconds*

Table 9 - 14 User Group: Distribution

Intercept Pilot ID	
System entity providing the audio prompts for calls holding in the queue.	None * User ID User Group ID AutoAttendant ID Voice ID

* = default settings

The Intercept Pilot should be set to a voice ID of the IVP card. This voice ID will have an intercept point defined (see “Setup Voice ID with Intercept Pilot” - section 9.7.2).

9.7.4 System default message

The system default message is a prerecorded system message for callers in the Hold Queue. The message played is, "All agents are currently busy. There are x calls ahead of you. To continue waiting, please stay on the line. To leave a message for this extension, press the 'pound' key." The x represents the caller's point in queue or the number of queued calls ahead of the caller.

Callers are able to stay on the line, hang up, or press # to leave a message. The message will repeat itself at the time interval defined in the user or user group Intercept Period.

Note
For system default messages only, the Intercept Point Dial Node ***must*** be a Dial Node under a Get Extension Node for proper operation of the "press #" feature.

9.7.5 Custom messages

While custom messages cannot take advantage of the point-in-queue indication, they should provide reassurance to callers, letting them know that the system is aware that they are still holding.

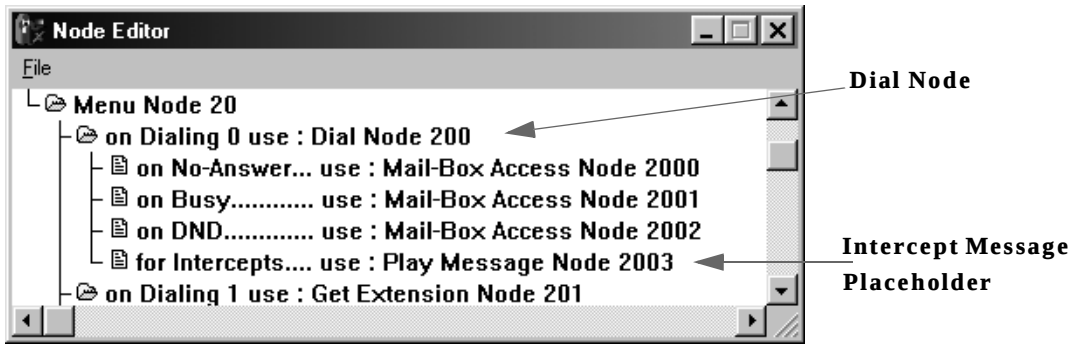
For a custom message to be played, the Intercept Message Placeholder must be redefined and the user or user groups mailbox must be configured to allow for custom messages to be played for queued calls to the user or user group.

9.7.5.1 Define the Intercept Message Placeholder

A Play Message Node or a Menu Node must be defined in the Intercept Message Placeholder under the Intercept Point's defined Dial Node.

A. Play Message Node

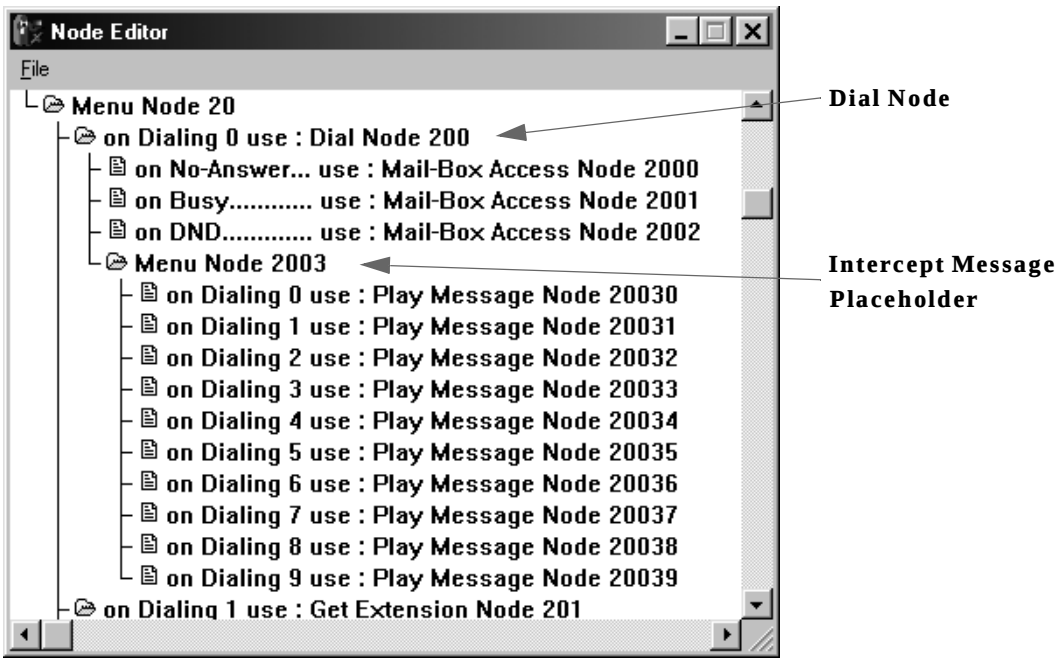
A Play Message Node will play the recording associated with the node for the caller. The Play Message Node will ignore any digits dialed by the caller and return the call to the held state once the message is through playing.



To properly configure the Play Message Node, the Play action for the node should be set to Play Recorded Message and a message should be recorded for the node. The node will ignore the setting Next Node and return the call to a held state after the message has played. For the steps to modify the Play Message Node properties, see "Modifying node properties" - section 9.3.14.

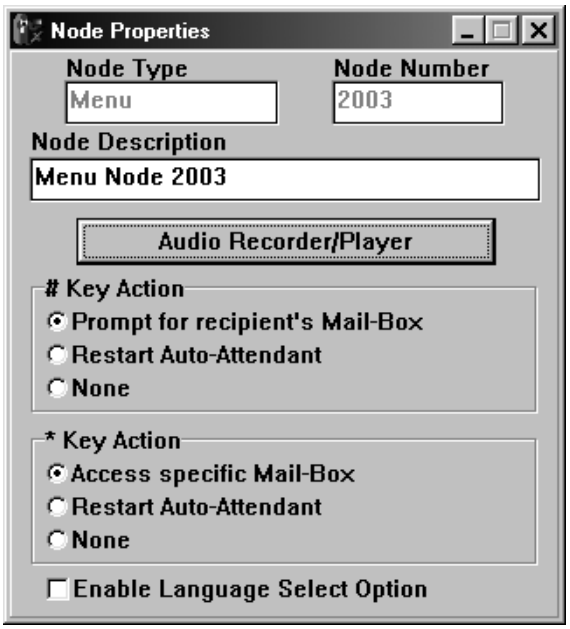
B. Menu Node

A Menu Node will play the recording associated with the node for the caller. The caller will have the option to press digits that have been mapped under the Menu Node to hear prerecorded messages or stay on the line to continue holding for the user or user group. The nodes defined under a Menu Node can only be Play Message Nodes and More Menu Nodes. All other types of nodes will not operate in an Intercept. See “Adding a Menu Node” - section 9.3.10.1.



To properly configure the Menu Node, The choices for the caller entering a * or # must be defined. Any menu choices under the Menu Node must also be defined. See “Adding a Menu Node” - section 9.3.10.1 and “Menu Node Properties” - section 9.3.15.1.

Note
All nodes defined under a Menu Node in an Intercept must be Play Message Nodes or Menu Nodes, other types of nodes will not operate properly in an Intercept.



9.7.5.2 Allow custom message playing.

For the system to play a custom messages, the user or user groups mailbox must be configured to allow for custom messages to be played for queued calls to the user or user group. To modify the mailbox for the user or user group that is being set up to have custom messages played for queued calls, follow the following steps:

1. Log in to the user or user group mailbox. This can be done by accessing any mailbox to leave a message and dialing * and the desired mailbox to configure.
2. Enter the mailbox password (if applicable, default is no password).
3. Press **'3'** to modify mailbox parameters.
4. Press **'8'** ten times to forward to parameter "set busy options".
5. Press **'1'** to modify this parameter.
6. Press **'2'** to allow for custom intercept messages to be played.
7. When the system states "Your entry has been saved", hang up.

This setting can also be modified through the VPS Mail-Box Editor of the Plexus Administrator Software. See “Voice Mailboxes” - section 9.4 for information.

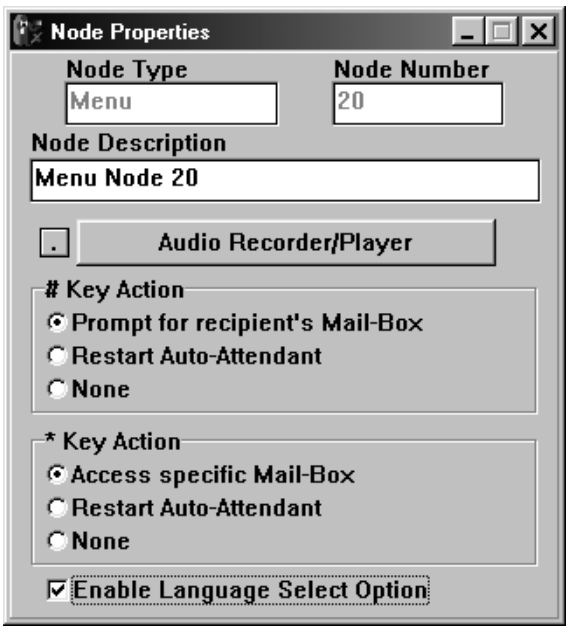
9.8 Multi-Language Option

9.8.1 Description

The Multi-Language Option allows incoming callers to select a language in which the menu options will be announced. Mailbox owners may select the language in which their mailbox options are read.

Note
You must have a multi-language enabled Integrated Voice Processor (IVP) Interface card for this option to be implemented.

9.8.2 Setup



Enable Language Selection Option	
Enables language selection option	Enable / disable*

* = default setting

T-1 / E-1 CAS

10

10.1 Overview

Channel Associated Signaling (CAS) is a generic name for a signaling method used on T-1 and E-1 carriers. While there are clear differences in the signaling methods used on the two carriers, they do both provide signaling information on a per-channel basis.

T-1 uses Robbed Bit Signaling (RBS). This method utilizes all 24 channels for voice, data, and signaling. Signaling between the central office and the customer equipment is accomplished through a combination of in-band DTMF/MF tones and bit robbing. Depending on the framing format being used (D4 or ESF), the least significant bits of certain frames are robbed and used for signaling. Framing is accomplished by inserting framing bits between each frame.

E-1 provides 30 "clear" channels for voice and data. The two remaining channels are reserved for framing and signaling. Despite the appearance of a common signaling channel, E-1 still qualifies as CAS because the signaling channel sends channel associated signaling bits.

CAS Carriers Compared		
	T-1	**E-1**
Total Channels	24	32
Usable Channels	24	30
Total Bandwidth	1.544 Mbps	2.048 Mbps
User Data Rate	1.536 Mbps	1.920 Mbps
Framing Method	in-band framing bits	dedicated channel (time slot 0)
Signaling	in-band DTMF/ tones and bit robbing	in-band MFC/DTMF tones and dedicated channel (time slot 16) for supervision bit signaling

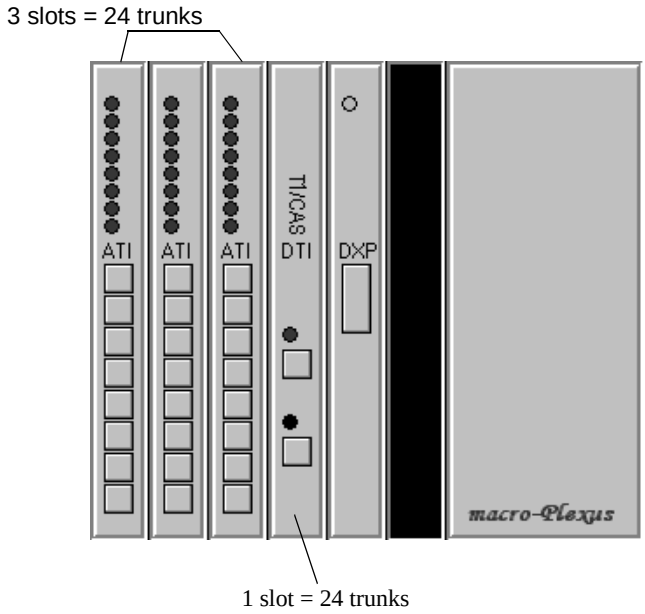
This chapter covers installation, configuration, and troubleshooting for both the T-1 and the E-1 digital trunk interfaces. The chapter includes the following sections:

Features and Advantages	A brief discussion of the main features and advantages of T-1 and E-1.
Hardware Description	A description of the hardware, including connectors, LEDs, and the applicable specifications.
Installation	Instructions on installing the card and connecting it to the carrier.
Configuration	Instructions on configuring the card and detailed explanations of the configurable parameters.
Troubleshooting	Instructions on interpreting the real-time status indicators and using the diagnostics available in Plexus Administrator.
Glossary of Common Terms	Definitions of many of the terms used in this chapter. Look here for detailed information about concepts you may encounter when ordering, installing, or configuring your T-1 or E-1 carrier.

10.2 Features and Advantages

The primary incentive for using T-1 or E-1, rather than analog POTs, is cost reduction. In many markets, these digital carriers are less expensive than the equivalent number of POTs lines.

Another advantage of T-1 and E-1 is that they allow users to make more efficient use of switch resources. Slots 8 and 10 in the Plexus system cabinet are designed to accommodate up to thirty channels each, to a total of sixty channels. Therefore, thirty trunks can be made available to a system using a single slot in the cabinet. By comparison, 4 slots would be needed to make this many trunks available using analog, 8-port CO cards.



T-1 makes more efficient use of the available card slots.

Note
In the case of T-1, one slot could provide up to 24 trunks to the system. Three slots would be needed using analog, 8-port CO cards. An E-1 could up to 30 trunks to the system. Four slots would be needed using analog, 8-port CO cards.

While the incentives for using T-1 or E-1 for voice transmission are mostly economic, there are also call routing and billing features that may prove valuable in some applications.

ANI and DNIS are often available on T-1 and E-1 carriers. The information provided by these services can be used to automatically route calls to specific users or user groups on the system, bypassing the auto-attendant. For example, you can route all callers that dial one (or more) of your assigned directory numbers to the sales group and those that dial a different number(s) to the tech support group.

Note
ANI and DNIS information is sent in-band (on the voice channels) on T-1 and E-1 carriers. MFC signaling is used on E-1 and DTMF tones are used on T-1. The format of the DTMF string sent on a T-1 is *ANI*DNIS*.

Calls made on an E-1 can be tagged by the Plexus system with an ID known as the Outbound Caller-ID Tag. Each user on the system can have their own ID. If the IDs correspond to legitimate directory numbers for the E-1 they may be sent as the Caller ID information[1]. The Outbound Caller-ID tag can also be used for call reporting. Many providers record the Outbound Caller ID data and present it on their billing statements. This allows customers to track their expenses on a user-basis.

1. Check with your provider.

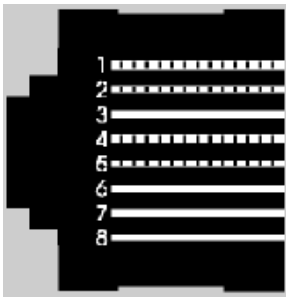
10.3 Hardware Description

The T-1 and E-1 peripheral cards are available with support for one full or fractional span.

All of the resources necessary to connect directly to the network interface are provided on the cards. For example, the CSU/DSU is fully-integrated on the T-1 card. The use of additional equipment, such as an external CSU or channel bank is not required and could complicate system setup.

10.3.1 Connectors

The T-1 and E-1 peripheral cards include up to two 8-pin connectors wired to the RJ48X standard (EIA TIA 568B compatible). The pin out of the connectors is as follows:

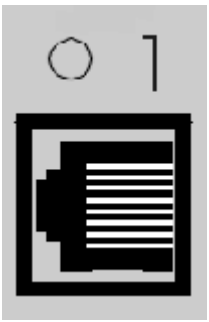


RJ48X connector

RJ48X Connector Pin Out	
Pin	**Signal**
1	Rx Tip
2	Rx Ring
4	Tx Ring
5	Tx Tip

10.3.2 LEDs

A red LED is provided for each span. Five different states provide status information as follows:



Connector and LED

Flash Cycle (in seconds)	Status
1/4 ON, 1/4 OFF	Functioning, but not receiving or misconfigured (i.e., no frame alignment)
1 ON, 1 OFF	OK
2 ON, 2 OFF	In loopback mode (see “Line LB” and “Payload LB” on page 25) Only applies to T-1.
2 X 1/4 ON, 1/4 OFF, 1 X 1/4 ON, 3/4 OFF (3 rapid flashes, pause)	Not properly communicating with the digital switch processor (DXP). This can happen if the loaded XOS version is not compatible with digital trunks
OFF	Not functioning

10.3.3 Specifications

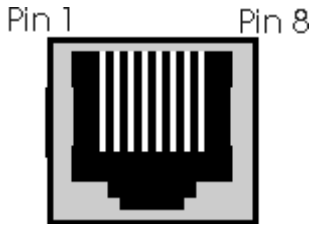
ITU-T I.431
ITU-T G.703 and G.704
ITU-T Q.930 (I.450) and Q.931 (I.451)

10.4 Installation

Carrier installation is the responsibility of the provider. After the customer places an order, the provider sends out technicians to bring the necessary cabling into the customer premises and terminate it near the PBX. The location where the carrier is terminated is called the network interface or demarcation point. The connection available at the demarcation point may be a single 8-pin connector or two coaxial connectors, one for transmit and one for receive. The customer is responsible for the connection between the demarcation point and the card.

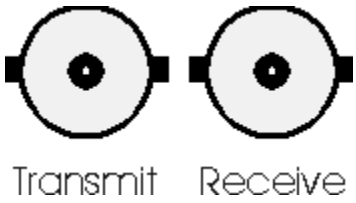
Demarcation

T-1 Carrier



8-pin

E-1 Carrier



Coaxial

A device provided with the E-1 card converts the coaxial demarcation into an 8-pin connector with the correct four wires terminated.

Coaxial to 8-pin converter

Note
The two coaxial connectors on the converter device are labeled "Tx" for transmit and "Rx" for receive.

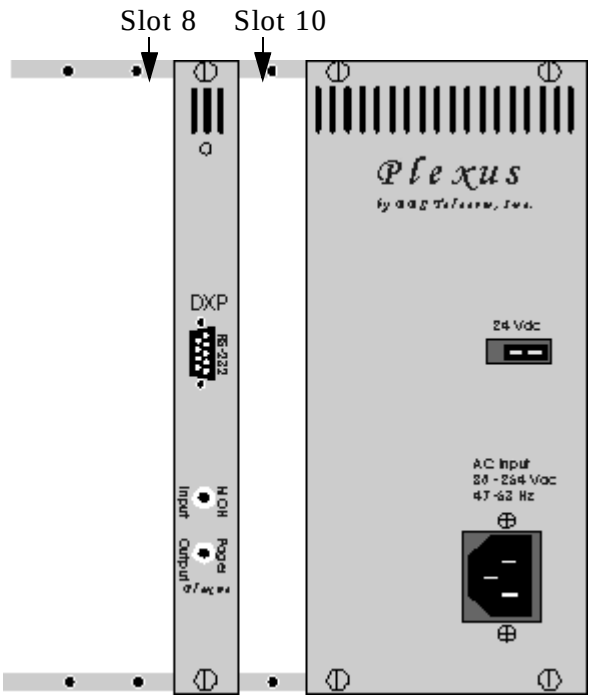
10.4.1 Inserting the card

1. Power off the system.
2. Carefully remove the card from the packaging.

Note
The cards are sensitive to static electricity and should be handled by the edges only. Never touch the components on a card.

3. Place the edges of the card inside the top and bottom card guides of either slot 8 or slot 10.

Note
If a T-1 or E-1 is installed in slot 10, slot 8 may only contain another T-1 or E-1 or an Inter-Unit Link Interface (ILI) card. If the T-1 or E-1 is installed in slot 8, no similar limitations exist.



T-1 and E-1 cards can only be inserted into slot 8 or slot 10. These slots support a larger number of channels than other slots in the system cabinet.

4 Gently slide the card in until it completely seats into the backplane. There will likely be an increase in pressure and a slight noise as the card is seated.

10.4.2 Making the connection

Tip

For straight through runs of less than 10 feet, a standard 10 BASE T cable can be used. The pin out discussed in the “Connectors” section (1 + 2 on one twisted pair and 4 + 5 on another), should be used for longer runs.

Using a cable that conforms to the following wiring diagram, connect the card to the demarcation point (or the converter device, if applicable). Either RJ-11 or RJ-45 plugs can be used.

Card-to-Network Interface Straight Through	
1	1
2	2
4	4
5	5

Card-to-Card (Tie-Line Application) Crossover	
1	5
2	4
4	2
5	1

1 Insert one end of the cable into one of the connectors on the card.

2 Insert the other end into the connector installed by the provider (i.e., the demarcation point) or into the converter device as illustrated below.

Inserting cable into converter device

Note
If you move the Plexus system further away from the demarcation point than it was at the time of installation, you may need to adjust the line build-out setting.

10.4.3 Activating service

As soon as the card is connected to the demarcation point, you are ready to activate service. When you contact the provider to do this, make sure that you take notes. There are several configurable parameters on both T-1 and E-1 carriers and it is extremely important that you and the provider select all the same options. If the same options are not implemented on both ends, the carrier will not function properly.

In some cases, the options selected are up to the customer. Therefore, you may want to review the glossary beginning on page 27 before activating service. This will allow you to make informed decisions and select the options that are best suited for your customer's application.

The following checklist highlights several of the configurable parameters you should discuss with the provider. You may want to circle the options selected for future reference.

Parameter	Options	
	T-1	**E-1**
Line Code	B8ZS AMI	HDB3 AMI
DID Size	1-10 digits	1- 10 digits
Framing Format	ESF D4	N/A
CRC-4 Frame Align	N/A	Yes No
Circuit Signaling	E&M - Wink Start E&M - Delay Start E&M - Immediate Start Loop Start Ground Start	N/A
Signal Type	N/A	MFC DTMF The E-1 card only supports R2 signaling. R1 is not supported.
Register Signaling	N/A	Brazilian Signaling ITU Generic Korean Signaling TELMEX Signaling Other[a]
MFC Timing	N/A	Max ON = Min Cycle =
Line Build-Out	Ask installers for code.	N/A
Flash Period	milliseconds	milliseconds
DTMF Timing	On = Off =	On = Off =

Parameter	Options	
Main Directory Number		
Other Directory/DID Numbers		
ANI	Yes No	Yes No
DNIS	Yes No	Yes No
Raw-DID enabled	Yes No	N/A

a.Contact BBS Telecom for assistance in developing a register signaling profile. Exact specifications from the carrier will be required

10.5 Configuration

Configuration is necessary to set parameters of the card so that it may properly communicate with the carrier. The parameters must be correctly configured in order to utilize the carrier.

Configuration is accomplished using the Plexus Administrator software utility. After launching the software and completing the peripheral card layout (see the Software Configuration section of the Plexus System Manual), you can begin configuring the T-1/E-1 card, as follows:

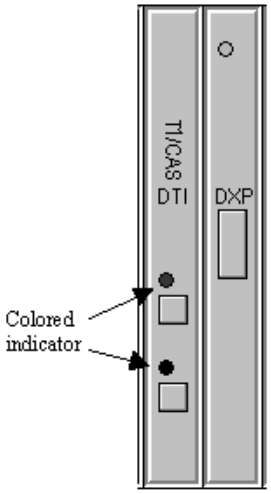
1. Click on the port on the image of the card.
2. Click on each tab and address each parameter in the order that they are presented.

Below, you will find an explanation of each parameter, the available options, and an indication of the default setting. As you set the parameters, refer back to the checklist to determine which settings were selected.

Note
It is extremely important that the same options be implemented on both ends of the T-1 / E-1.

10.5.1 The T-1 / E-1 Peripheral Card

Before clicking on the port on the image of the card, you will notice that the port has a colored indicator. Each color of the indicator represents a different state for the T-1 / E-1 span.



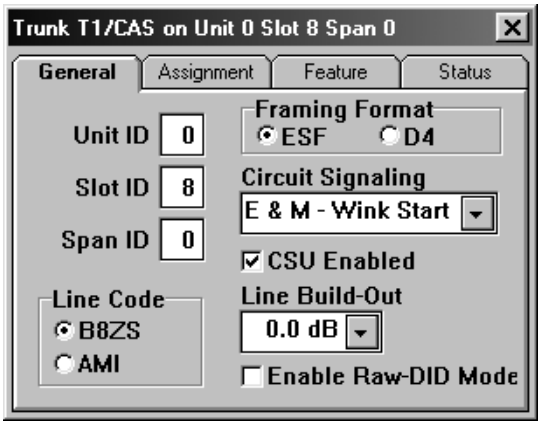
Red – No channels have been mapped to trunks.
Green – At least 1 channel has been mapped to a trunk and all mapped channels are enabled.
Yellow – At least 1 channel that has been mapped to a trunk has been disabled.
Black – The span is not active / available

Note
The Plexus Administrator software will show two spans on each T-1 / E-1 peripheral card. The second span will always show not active / available. Ensure that all programming be completed on the top span in the Plexus Administrator software. The second span shown is for future release of two span T-1 / E-1 peripheral cards.

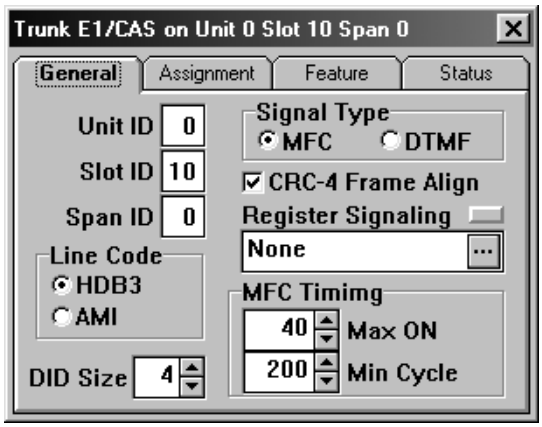
10.5.2 General

The General tab includes system ID information and most of the carrier-specific configurable parameters. Selections are made by clicking on the appropriate radio button or pull-down menu option. Values can be increased or decreased, within the allowable range, using the arrow keys on your keyboard or by clicking on the appropriate arrow image.

T-1 Carrier



E-1 Carrier



This version of the General window will appear when "Enable Raw-DID Mode is selected.

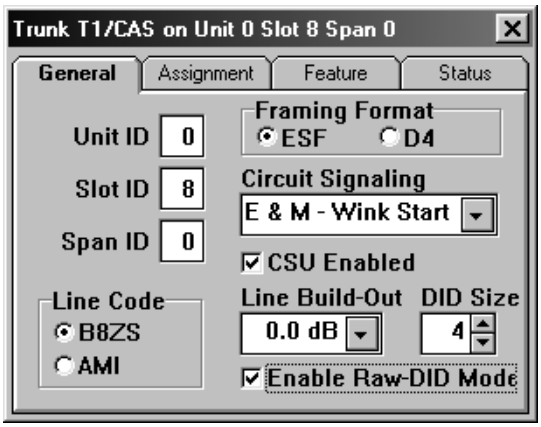


Table 10 - 1 T-1/E-1 CAS: General

Unit ID	T-1	E-1
Logical identifier for the system being configured. Unless there is more than one system, as in a linked environment, this ID will be 0.	Automatically Assigned	Automatically Assigned
Slot ID		
Logical identifier for the slot in which the card is inserted.	Automatically Assigned	Automatically Assigned

Table 10 - 1 T-1/E-1 CAS: General

Span ID		
Logical identifier for the span being configured.	Automatically Assigned	Automatically Assigned
Line Code (Glossary page 31)		
Method of manipulating the bit pattern to maintain synchronization between the central office and the card.	B8ZS* AMI	HDB3* AMI
DID Size (Glossary page 27)		
The number of digits of the dialed number that the central office will send with a call.	4* Range: 1-10	4* Range: 1-10
Framing Format (Glossary page 29)		
Method of organizing the bit pattern so that the receiving end can identify the channel (time slot) from which the data was sampled.	ESF* D4	N.A.
Signal Type (Glossary page 32)		
The type of in-band signaling. MFC is the standard in-band signaling used on E-1 card-to-network interface applications. The selected option applies to all channels on the span.	N.A.	MFC* DTMF
CRC-4 Frame Align		
Instructs the card to send cyclic redundancy check (checksum) information to the central office.	N.A.	Disabled* Enabled
Circuit Signaling (Glossary page 27)		
The type of line interface to emulate. The selected option applies to all channels on the span.	E & M - Wink Start* E & M - Delay Start E & M - Immediate Start Loop Start Ground Start	N.A.

Table 10 - 1 T-1/E-1 CAS: General

Register Signaling (See "Signal Type" on page 32.)		
The MFC signaling profile that will be used. New profiles can be generated by double-clicking on any of the selections and modifying the category/tone relationship.	N.A.	None* Brazilian Signaling ITU Generic Korean Signaling TELMEX Signaling
CSU Enabled		
Defines if the card is acting as the CSU or if there is an external CSU.	Enabled* Disabled	N.A.
Line Build-Out (Glossary page 30)		
The amount of line build-out (LBO) necessary to attenuate signals transmitted from the card so that they are properly received and handled by receiving equipment.	0.0 dB* -7.5 dB -15.0 dB -22.5 dB	N.A.
MFC Timing - Max ON (See "Signal Type" on page 32.)		
The maximum allowed duration for MFC signals sent on the E-1 carrier. A certain duration may be mandated by the provider.	N.A.	40 sec* Range: 1-255 sec
MFC Timing - Min Cycle (See "Signal Type" on page 32.)		
The minimum allowed duration for a forward/backward MFC cycle. A certain duration may be mandated by the provider.	N.A.	200 msec* Range: 1-1000 msec
Enable Raw-DID Mode		
Enables the T-1 to act as a tie line between two systems or to accept digits in a non-standard form from the service provider. ***NOTE***: ANI and DNIS digits should sent in the format *ANI*DNIS*. If your provider is only sending DID with no ANI or *s (or any set of raw digits), select this mode. If this mode is enabled, ANI/DNIS capture is unavailable.	Enable Disable*	N.A.

* = default setting

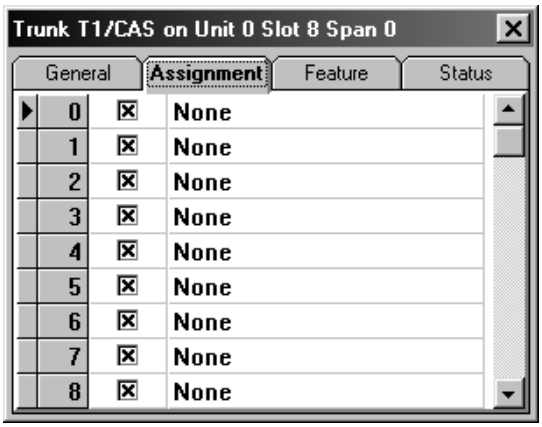
10.5.3 Assignment

The assignment: channel association tab includes checkboxes for enabling and disabling channels and pull-down menus for assigning trunks to channels. Channels are enabled and disabled based on the number of channels provided with the span.

Tip

To quickly assign a range of trunks to the enabled channels, right-click anywhere in the third column and select AutoAssign. To quickly enable / disable channels, right-click anywhere in the second column and choose Auto Assign to keep only assigned channels enabled, enable ALL channels, or disable ALL channels.

T-1 Carrier



E-1 Carrier

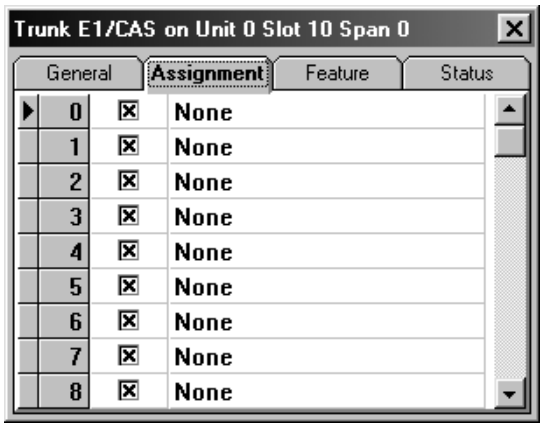


Table 10 - 2 T-1/E-1 CAS: Assignment

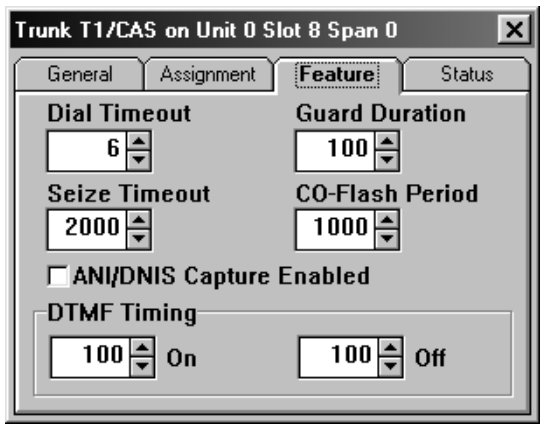
Channel ID	T-1	E-1
Channel identifier.	0 - 23, 23*	0 - 29, 29*
Checkbox		
Enables/disables the corresponding channel. All channels should be enabled for full spans. A subset of the channels is enabled for fractional spans.	Enabled* Disabled	Enabled* Disabled
Trunk Assignment Pull-Down		
The pull-down menu is used to assign a trunk to each channel. A 1-to-1 relationship must exist between trunks and channels.	None* Trunk ID	None* Trunk ID

* = default setting

10.5.4 Feature

The feature tab includes a checkbox to enable ANI and DNIS capture and several programmable timing parameters. The timing parameters can be adjusted, within the allowable range, using the arrow keys on your keyboard or by clicking on the appropriate arrow image.

T-1 Carrier



E-1 Carrier

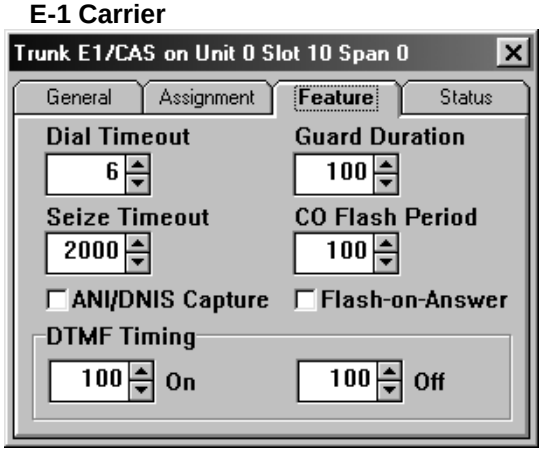


Table 10 - 3 T-1/E-1 CAS: Feature

Dial Timeout	**T-1**	**E-1**
The period of time that must elapse before the system assumes that the user is finished dialing.	6 sec* Range: 0-255 sec	6 sec* Range: 0-255 sec
Guard Duration		
The period of time that a channel is unavailable for placing an outbound call upon completion of a previous call. The guard duration prevents call collisions.	100 msec* Range: 0-65535 msec	100 msec* Range: 0-65535 msec
Seize Timeout		
The period of time that the system will wait when attempting to seize a channel to make an outbound call.	200 msec* Range: 50-2000 msec	200 msec* Range: 50-2000 msec

Table 10 - 3 T-1/E-1 CAS: Feature

CO Flash Period		
The on-hook duration that the central office interprets as a flash signal. This timing parameter varies from one central office to another.	1000 msec* Range: 50-2000 msec	100 msec* Range: 50-2000 msec
ANI/DNIS Capture Enabled		
Instructs the card to look for ANI and DNIS information on incoming calls. If you are receiving ANI and/or DNIS on your T-1, this parameter should be enabled. ***NOTE:*** The ANI/DNIS digits should be sent in the standard format *ANI*DNIS*. If ANI or DNIS is not available, the digits should be sent as **DNIS* or *ANI**, respectively. If Enable Raw-DID Mode is enabled, this selection will be unavailable.	Disabled* Enabled	Disabled* Enabled
Flash-on-Answer		
Used to "flash" the CO on seizing an inbound call. This is used to signal the CO to reject certain types of calls (e.g., collect calls).	N.A.	Disabled* Enabled
DTMF Timing		
The duration of a regenerated DTMF digit (On) and the interdigit period (Off) when dialing a string.	100 msec* 50-255 msec	100 msec* 50-255 msec

* = default setting

10.5.5 Call Routing

In order to use ANI and DNIS information to route calls, you must first establish a relationship between each number and the user, user group, or other system entity to which you want the call routed. These relationships are defined on the system's route map. See System Parameters: Route Map in the Software Configuration section of the Plexus System Manual.

If you obtain a large enough block of directory numbers from the provider, a unique number can be assigned to each user on the system. Using the DNIS information and an assignment on the system's route map, calls can then be directly routed to users based on the number dialed, as on a DID trunk. If the user's assigned directory number corresponds with their Outbound Caller-ID tag, return calls can be placed to the user based on the Caller ID [E-1 only].

Note
Check with your provider about getting additional directory numbers. Typically, directory numbers are available in blocks.

10.5.6 Outbound Caller-ID tag

An Outbound Caller-ID tag should be defined for each user that will be placing calls on an E-1 [see User: Feature in the Software Configuration section of the Plexus System Manual]. If the Outbound Caller-ID tags correspond to legitimate directory numbers, the number is typically forwarded on outbound calls as the Caller ID[1]. If the Outbound Caller-ID tags are not directory numbers, they can still be used to track calls on a user-basis as many providers record the Outbound Caller ID data and present it on their billing statements.

1. Check with your provider.

10.6 Status Monitoring & Troubleshooting

Plexus Administrator provides an interface which can be used to monitor the real-time status of your installed and configured carriers. In the case of T-1, real-time diagnostics are provided to assist in troubleshooting.

To monitor your carrier, proceed as follows:

1. Launch Plexus Administrator.
2. Open the appropriate configuration file.
3. Select **Open** from the Link menu.
4. Click on either port on the image of the card.
5. Click on the **Status** tab.

10.6.1 Status

The status tab includes a status indicator for each installed span. In the case of T-1, several indicators provide real-time diagnostic information about the state of both the received and the transmitted signal.

T-1 Carrier

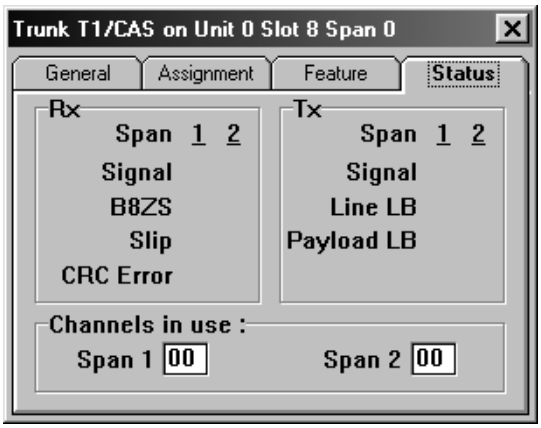


Table 10 - 4 T-1 CAS: Status - Receive (Rx)

Signal	
Indicates the state of the received signal on the corresponding span. Different colors indicate different states.	**Green** = properly framed signal, no alarm conditions. **Yellow** = properly framed signal, alarm condition. Alarm conditions can occur for several reasons, such as a loss of signal defect or during maintenance. **Blue** = unframed. During maintenance, an unframed signal may be received. **Red** = no signal.
B8ZS	
Indicates that a B8ZS bit pattern was detected on the corresponding span. If you are using AMI line code, no indication should ever be displayed. If an indication is received in such a case, you should check to make sure that the carrier is, in fact, using AMI.	**Red** = B8ZS bit pattern detected.
Slip	
Indicates that a received frame was either replicated or deleted. This can occur when there is a difference between the timing of a synchronous receiving terminal and the received signal.	**Red** = Slip error detected.
CRC Error	
Indicates that a received cyclic redundancy checksum (CRC) is not identical to the corresponding locally-calculated checksum. CRC Errors are only detected when using the ESF framing format.	**Red** = CRC error detected.

Table 10 - 5 T-1 CAS: Status - Transmit (Tx)

Signal	
Indicates the state of the received signal on the corresponding span. Different colors indicate different states.	**Green** = properly framed signal, no alarm conditions. **Yellow** = properly framed signal, alarm condition. **Blue** = unframed.
Line LB	
Indicates that the corresponding span is currently in line loopback mode. The provider initiates a loopback to perform diagnostic testing on a T-1 carrier. A line loopback results in a full 1.544 Mbit/s loopback of the signal received at the card.	**Red** = line loopback in progress.
Payload LB	
Indicates that the corresponding span is currently in payload loopback mode. A payload loopback results in a 1.536 Mbit/s loopback of the signal received at the card, maintaining bit-sequence integrity. Payload loopbacks can only be performed on a T-1 using the ESF framing format.	**Red** = payload loopback in progress.
# Channels in use	
This shows the number of channels currently in use on the T-1. This can be used to monitor system call load or when doing system maintenance.	0-23

E-1 Carrier

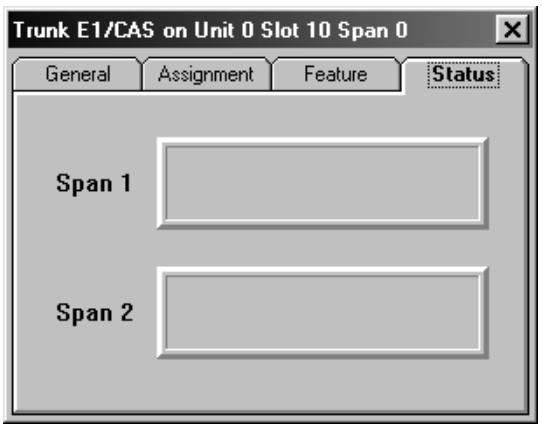


Table 10 - 6 E-1 CAS: Status

Span #	
Indicates the state of the signal on the corresponding span and the number of channels currently in use. Different colors indicate different states.	**Green** = good signal **Red** = no signal

10.7 Glossary

10.7.1 ANI

Automatic Number Identification; the caller's phone number (per directory listing)

10.7.2 Circuit Signaling

Circuit signaling refers to the type of line interface that the T-1 will emulate. The Plexus T-1 card is capable of emulating the following line interfaces:

- E&M – Wink Start
- E&M – Delay Start
- E&M – Immediate Start
- Loop Start
- Ground Start

The type of circuit signaling you select depends on the signaling information required by your application.

10.7.3 Clear

Indicates that only user data is on these channels; no bits are robbed or stuffed for signaling or framing.

10.7.4 DID

See DNIS.

10.7.5 DID Size

The DID Size indicates the number of digits of the dialed number that the central office will send on an incoming call. For example, if the dialed number was 328-9500 and the DID Size was 4, the central office would send 9500 (i.e., the numbers at the end of the string).

The delivery of the dialed number information is also known as Dialed Number Identification Service (DNIS). On an E-1, this information is sent using the forward/backward methodology of MFC signaling. In the example above, the receiving end would indicate that it had all the numbers that it needed after receiving the second "0." At this point, it would request that the central office send any caller ID information.

When activating service, make sure to ask your provider what DID Size is applicable on your T-1 / E-1 carrier.

10.7.6 DNIS

Dialed Number Identification Service; the number that the caller called.

10.7.7 E&M

The E&M interfaces are the simplest and most commonly used on T-1 carriers. E&M can communicate only two states, on-hook and off-hook, between the transmitting and receiving ends.

The differences between the three supported E&M interfaces has to do with channel seizure. Wink start requires a "wink" from the receiving end to acknowledge seizure before any further signals can be sent from the transmitting device. Delay start requires a wink followed by a fixed delay before the transmitting device can proceed. The delay gives the receiving device time for set up. Immediate start requires neither a wink nor a delay. The transmitting device can immediately send signals such as DTMF upon channel seizure. Immediate start is not frequently used because of the lack of handshaking signals which can cause signaling to be lost and a higher incidence of mis-routed calls.

The signaling on all of the supported E&M interfaces is symmetrical between the transmitting and receiving ends. E&M interfaces give clear answer/ disconnect supervision.

Note
Plexus tie-line applications must use one of the supported E&M interfaces. Loop start and ground start are only supported for T-1 card-to-network interface applications.

10.7.8 Fractional T-1/E-1

A span with less than the full number of channels enabled.

10.7.9 Framing Format

All of the bits transmitted on a T-1 carrier are meaningless unless the equipment at the receiving end knows how they are organized. Framing is what supplies organization to the bits. The Plexus T-1 card supports two common framing formats, superframe (D4) and extended superframe (ESF).

A frame contains an 8-bit sample from each of the time slots or channels, plus a single framing bit (193 bits in total). The framing bit allows the receiving equipment to identify the beginning of a frame.

The D4 format groups frames into blocks of 12. These blocks are called superframes. The framing bits in each superframe repeat a defined sequence that the receiving equipment uses to locate the framing bit. Signaling is accomplished by 'robbing' the least significant bit in the 6th and 12th frame (hence the name, Robbed Bit Signaling) from each channel in the superframe. By robbing two bits from each channel, four different states may be represented: 00 (on hook), 01, 11 (off hook), and 10.

The extended superframe (ESF) format improves upon the D4 format in several ways. First, it doubles the number of frames in the superframe from 12 to 24. This enables two more bits (in frames 18 and 24) to be used for signaling thereby expanding the states that may be represented. Second, ESF only uses every fourth framing bit for synchronization (6-bit framing pattern). This leaves 6,000 bit/s for new functions, primarily diagnostics and statistics. 2,000 of these bits are used for continuous error checking and the remaining 4,000 are used by the network for performing loopbacks and measuring various statistics.

In applications where only voice is being transmitted, D4 is adequate. For applications involving the transfer of large amounts of modulated data (i.e., using modems), ESF is the better choice. As is the case with line code, the most important consideration is that the framing format be the same on both ends. Therefore, make sure that you and your provider agree on the selected option.

10.7.10 Glare

Collisions resulting from an attempt to seize a channel on which an inbound call is being received.

10.7.11 Ground Start

The ground start option, like the loop start option, is not symmetrical. The network side of a ground start T-1 carrier can signal loop current feed, open loop current, and ringing. The card side can signal on-hook, off-hook, and service request. Disconnect supervision is available with ground start signaling and protection from glare is provided. ANI and DNIS are not available.

10.7.12 HDB-3 and B8ZS compared

The primary difference between HDB-3 (used on E-1) and B8ZS (used on T-1) are the number of consecutive zeros they allow. HDB-3 replaces 4 consecutive zeros with a bipolar violation (successive pulses with the same polarity). B8ZS replaces a string of 8 zeros with a pre-defined bit pattern.

10.7.13 In-band vs. supervision bit

In-band signaling occurs on the 30 voice channels. Supervision bit signaling is the signaling that takes place on time slot 16.

Note
The Plexus E-1 card only supports R2 signaling. R1 is not supported.

10.7.14 Line Build-Out

A Line Build-Out (LBO) attenuates the signal transmitted by the card (T-1 carrier only). LBO simulates cable loss to ensure that network equipment can properly receive and handle the transmitted signal. There are all sorts of equations used to determine the LBO value that is needed to properly attenuate a signal. Fortunately, your provider's installation personnel are responsible for informing you as to which LBO code applies at your installation. This information is used to determine the option you select in Plexus Administrator, as follows:

Code	Setting	w/ additional cable loss of 7.5 - 13.0 dB	w/ additional cable loss of 15.0 - 20.5 dB
A	0.0 dB	0.0 dB	0.0 dB
B	-7.5 dB	0.0 dB	0.0 dB
C	-15.0 dB	-7.5 dB	0.0 dB

Additional cable loss is achieved by moving the Plexus system further from the demarcation point. Unless the system needs to be relocated, it is recommended that you go with the advised code and retain the current position of the system. Otherwise, you will need to measure cable loss and determine whether you need to modify the LBO setting.

Note
The -22.5 dB setting is typically not used and is reserved for special applications.

10.7.15 Line Code

The line code has to do with how the binary data is coded. The line code options were developed in response to different challenges identified in receiving and synchronizing on the data bit pattern. AMI (alternate mark inversion) manipulates the data by reversing the polarity of successive 1's or marks. This helps the receiving equipment to maintain a zero voltage reference point.

B8ZS (binary 8-zeros suppression) and HDB3 (High Density Bipolar 3-zeroes), both developed subsequent to AMI, deal with "ones density" - the frequency of 1's in the data bit pattern. Ones density is important because T-1 and E-1 devices synchronize by tracking the 1's. A long string of 0's increases the likelihood that a device will be unable to maintain synchronization. While today's equipment is better able to deal with a long string of 0's, B8ZS and HDB3 are still viewed as improved line code alternatives and are therefore recommended.

The most important consideration when selecting the line code option is that the same option be used on both ends. Therefore, make sure that you and the provider agree on the selected option.

10.7.16 Loop Start

The loop start option emulates the line interface found on a typical analog POTs line. Loop start signaling is not symmetrical with regard to the states communicated between transmitting and receiving ends. The network side of a loop start T-1 carrier can signal loop current feed and ringing. The card side can signal on-hook and off-hook. Disconnect Supervision, ANI, and DNIS are not available with loop start signaling and glare is not inhibited.

10.7.17 MFC Signaling

MFC signaling uses a standard set of tones to communicate between the E-1 card and the central office. Different countries and providers assign different messages to the tones. Therefore, it may be necessary to map the different messages to the tones and create a profile. Several pre-defined mappings are provided.

MFC signaling uses a forward (from the sending side) /backward (from the receiving side) methodology.

10.7.18 POTs

Plain Old Telephone Lines

10.7.19 Signal Type

The signal type option on an E-1 carrier refers to the in-band signaling on an E-1 carrier and is analogous to the circuit signaling option on a T-1. The choices are Multi-Frequency Compelled (MFC) or Dual Tone Multi-Frequency (DTMF). MFC is the required method for E-1 card-to-network interface applications in most countries. DTMF is used in some countries, but is more commonly used for tie-line applications.

10.7.20 Span

A group of channels or time slots.

10.7.21 Wink

A rapid off-hook, on-hook combination.

ISDN PRI

11.1 Overview

ISDN PRI is an interface between a user and a digital telephony network capable of delivering many types of services. The interface utilizes a set of communication protocols, including a message set used to request services. The user in the case of PRI is typically a PBX or some other switching equipment and the network is the central office (CO).

PRI can be implemented on either a T-1 or an E-1 carrier. In both cases, the full bandwidth of the bearer or B-channels is available for the transmission of voice and/or data. Signaling for the entire span is accomplished using a standard packet protocol on a single, common channel, called the D-channel. Therefore, PRI is often generically referred to as CCS, which stands for Common Channel Signaling.

PRI Carriers Compared		
	T-1	**E-1**
Total Channels	24 numbered 1-24	32 numbered 0-31
B-Channels	23	30
Total Bandwidth	1.544 Mbps	2.048 Mbps
User Data Rate	1.536 Mbps	1.920 Mbps
Framing Method		dedicated channel time slot 0
Signaling	D-channel time slot 24	D-channel time slot 16

This chapter covers installation, configuration, and troubleshooting for both versions of the Plexus ISDN PRI peripheral card. The chapter includes the following sections:

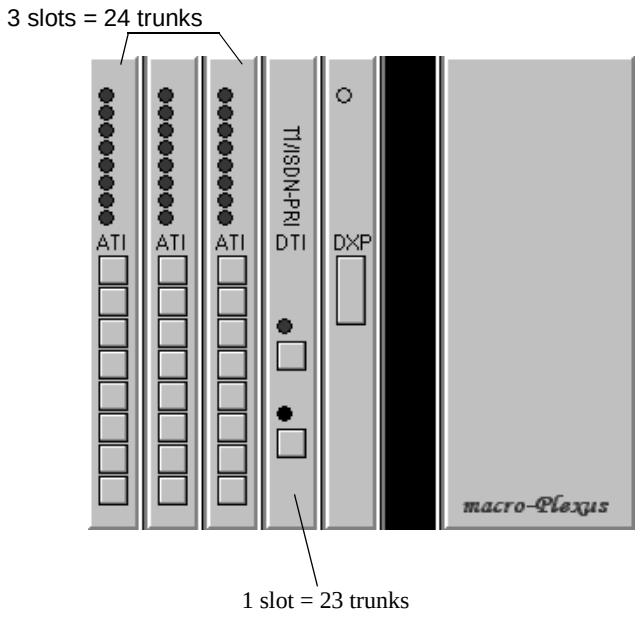
Features and Advantages	A brief discussion of the main features and advantages of ISDN PRI.
Hardware Description	A description of the hardware, including connectors, LEDs, and the applicable specifications.
Installation	Instructions on installing the ISDN PRI card and connecting it to the carrier.
Configuration	Instructions on configuring the card and detailed explanations of the configurable parameters.
Troubleshooting	Instructions on interpreting the real-time status indicators available in Plexus Administrator.
Glossary	A glossary of terms. Look here for detailed information about concepts you may encounter when ordering, installing, or configuring your ISDN PRI.

11.2 Features and Advantages

The primary incentive for using ISDN PRI, rather than analog POTs, is cost reduction. In many markets, an ISDN PRI is cheaper than the equivalent number of POTs lines.

Another advantage of ISDN PRI is that it allows users to make more efficient use of switch resources. Slots 8 and 10 in the Plexus cabinet are designed to accommodate up to sixty channels each; equivalent to two E-1 PRI spans. Therefore, sixty trunks can be made available to a system using a single slot in the cabinet. By comparison, 8 slots would be needed to make this many trunks available using analog, 8-port CO cards.

Note
If a T-1 or E-1 card is used in slot 10, slot 8 can only be filled by another T-1 or E-1 or by an Inter-Unit Link Interface card (ILI). Placing a T-1 or E-1 card in slot 8 produces no similar limitations.



ISDN PRI makes more efficient use of the available card slots.

Note
In the case of T-1, one slot could provide up to 23 trunks to the system. Three slots would be needed using analog, 8-port CO cards. An E-1 could up to 30 trunks to the system. Four slots would be needed using analog, 8-port CO cards.

While the incentives for using PRI for voice transmission are mostly economic, there are also call routing, billing, and other supplemental features that may prove valuable in some applications.

Calling Number Delivery and Called Party Number information are sent and received via message packets on the D-channel. This information can be used to automatically route calls to specific users or user groups on the system, bypassing the auto-attendant. For example, you can route all callers that dial one (or more) of your assigned directory numbers to the sales group and those that dial a different number(s) to the tech support group.

Calls made on a PRI can be tagged by the Plexus system with an ID known as the Outbound Caller-ID Tag. Each user on the system can have their own ID. If the IDs correspond to legitimate directory numbers for the PRI they may be sent as the Caller ID information[1]. The Outbound Caller-ID tag can also be used for call reporting. Many providers record the outbound Caller ID data and present it on their billing statements. This allows customers to track their expenses on a user-basis.

For information regarding support for supplemental services, check our web page at http://www.bbstelecom.com.

1. Check with your provider.

11.3 Hardware Description

The ISDN PRI peripheral cards are available with support for one full or fractional span.

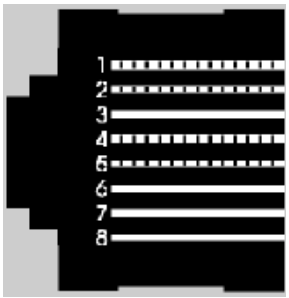
The ISDN PRI peripheral card in conjunction with the Plexus system provides all necessary signaling to the central office, distributes ISDN services to other devices that are attached to the system, and performs all protocol conversion functions for any attached analog devices. The ISDN PRI card therefore qualifies as NT2, or Network Termination Type 2, equipment.

All of the resources necessary to connect directly to a PRI network interface are provided on the card. No additional equipment is necessary.

Note
Unlike an ISDN BRI, a PRI is not typically connected to an end-user's terminal equipment, such as a phone or modem. Rather, PRI is used as a trunk to connect customer's switching equipment (i.e., PBX or other NT2 equipment) to the central office.

11.3.1 Connectors

The ISDN PRI cards include a 8-pin connector wired to the RJ48X standard (EIA TIA 568B compatible). The pin out of the connectors is as follows:

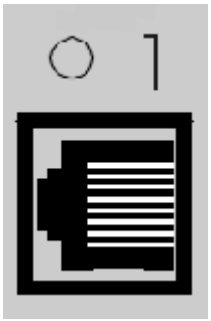


RJ48X connector

RJ48X Connector Pin Out	
Pin	**Signal**
1	Rx Tip
2	Rx Ring
4	Tx Ring
5	Tx Tip

11.3.2 LEDs

A red LED is provided for each span. Three different states provide status information as follows:



Connector and LED

Flash Cycle (in seconds)	Status
Flash: 1/4 ON, 1/4 OFF	Functioning, but not receiving or misconfigured (i.e., no frame alignment).
Flash: 1 ON, 1 OFF	OK
Off	Not functioning
Flash: 2 ON, 2 OFF	Not functioning - wrong software version on card

11.3.3 Specifications

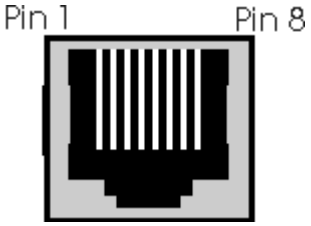
ITU-T I.431
ITU-T G.703 and G.704
ITU-T Q.930 (I.450) and Q.931 (I.451)

11.4 Installation

The installation of an ISDN PRI is the responsibility of the provider. After the customer places an order, the provider sends out technicians to bring the necessary cabling into the customer premises and terminate it near the PBX. The location where the PRI is terminated is called the network interface or the demarcation point. The connection available at the demarcation point may be a single 8-pin connector or two coaxial connectors; one for transmit and one for receive. The customer is responsible for the connection between the demarcation point and the PRI peripheral card.

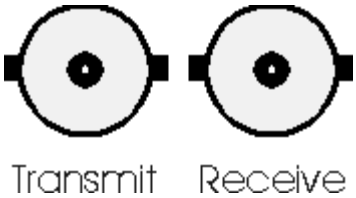
Demarcation

T-1 Carrier



8-pin

E-1 Carrier



Coaxial

A device provided with the E-1 version of the card converts the coaxial demarcation into an 8-pin connector with the correct four wires terminated.

Coaxial to 8-pin converter

Note
The two coaxial connectors on the converter device are labeled "Tx" for transmit and "Rx" for receive.

11.4.1 Inserting the card

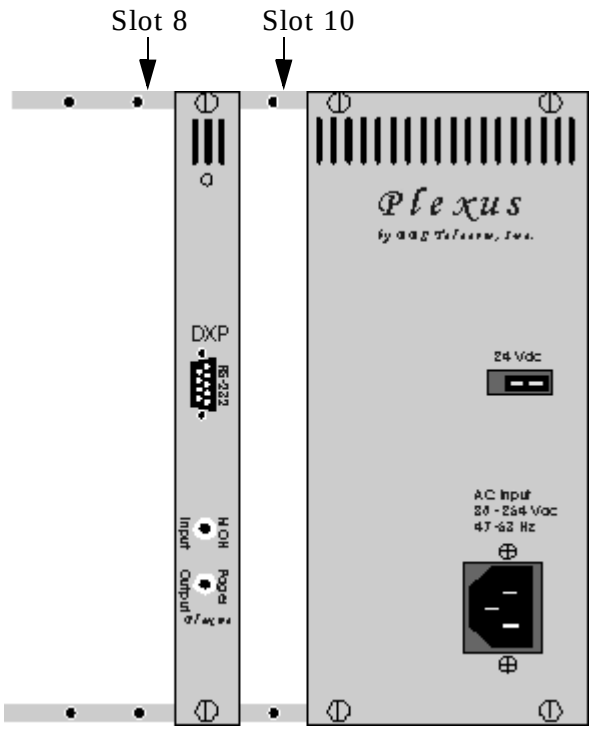
1 Power off the system.

2 Carefully remove the card from the packaging.

Note
The cards are sensitive to static electricity and should be handled by the edges only. Never touch the components on a card.

3 Place the edges of the card inside the top and bottom card guides of either slot 8 or slot 10.

Note
If a T-1 or E-1 is installed in slot 10, slot 8 may only contain another T-1 or E-1 or an Inter-Unit Link Interface (ILI) card. If the T-1 or E-1 is installed in slot 8, no similar limitations exist.



ISDN PRI cards can only be inserted into slot 8 or slot 10. These slots support a larger number of channels than other slots in the system cabinet.

4 Gently slide the card in until it completely seats into the backplane. There will likely be an increase in pressure and a slight noise as the card is seated.

11.4.2 Making the connection

Tip

For straight through runs of less than 10 feet, a standard 10 BASE T cable can be used. The pin out discussed in the “Connectors” section (1 + 2 on one twisted pair and 4 + 5 on another), should be used for longer runs.

Using a cable that conforms to the following wiring diagram, connect the card to the demarcation point (or the converter device, if applicable). Either RJ-11 or RJ-45 plugs can be used.

Straight Through	
1	1
2	2
4	4
5	5

1. Insert one end of the cable into one of the connectors on the card.
2. Insert the other end into the connector installed by the provider (i.e., the demarcation point) or into the converter device as illustrated below.

Inserting cable into converter device

Note
If you move the Plexus system further away from the demarcation point than it was at the time of installation, you may need to adjust the line build-out setting. See page 16.

11.4.3 Activating service

As soon as the ISDN PRI card is connected to the demarcation point, you are ready to activate service. When you contact the provider to do this, make sure that you take notes. There are several configurable parameters on a PRI and it is extremely important that you and the provider select all the same options. If the same options are not implemented on both ends, the PRI will not function properly.

In some cases, the options selected are up to the customer. Therefore, you may want to review the glossary in "Glossary" - section 11.7 before activating service. This will allow you to make informed decisions and select the options that are best suited for your customer's application.

The following checklist highlights several of the configurable parameters you should discuss with the provider. You may want to circle the options selected for future reference.

Parameter	Options	
	T-1	**E-1**
Line Code	B8ZS AMI	HDB3 AMI
CRC-4 Frame Align	N/A	Yes No
Terminal-Endpoint ID '0' is acceptable?	Yes No, use _____	Yes No, use _____
Switch Variant	US National ISDN-2 AT&T 4ESS AT&T 5ESS Nortel DMS-100	ETSI NET5 Australia TS014.1
Line Build-Out	Ask installers for code.	N/A
Main Directory Number		
Other Directory/DID Numbers		
Channel selection direction for inbound calls	Ascending (recommended) Descending	Ascending (recommended) Descending

Parameter	Options	
Dialing Method	En-bloc (recommended) Overlap	En-bloc (recommended) Overlap
Calling Number Delivery	Yes No	Yes No

11.5 Configuration

Configuration is necessary in order to set the various configurable parameters of the card so that it may properly communicate with the carrier. The parameters must be correctly configured in order to utilize the PRI.

Configuration is accomplished using the Plexus Administrator software utility. After launching the software and completing the peripheral card layout [see the Software Configuration section of the Plexus System Manual], you can begin configuring the card, as follows:

1. Click on the applicable port on the image of the card.
2. Click on each tab and address each parameter in the order that they are presented.

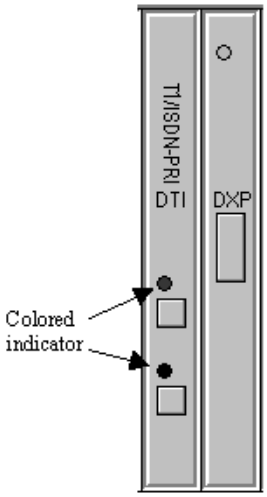
Below, you will find an explanation of each parameter, the available options, and an indication of the default setting. As you set the parameters, refer back to the checklist to determine which settings were selected.

Note
It is extremely important that the same options be implemented on both ends of the ISDN PRI.

.

11.5.1 The ISDN PRI Peripheral Card

Before clicking on the port on the image of the card, you will notice that the port has a colored indicator. Each color of the indicator represents a different state for the T-1 / E-1 span.



Red – No channels have been mapped to trunks.
Green – At least 1 channel has been mapped to a trunk and all mapped channels are enabled.
Yellow – At least 1 channel that has been mapped to a trunk has been disabled.
Black – The span is not active / available

Note
The Plexus Administrator software will show two spans on each T-1 / E-1 peripheral card. The second span will always show not active / available. Ensure that all programming be completed on the top span in the Plexus Administrator software. The second span shown is for future release of two span T-1 / E-1 peripheral cards.

11.5.2 General

The general tab includes system ID information and most of the carrier-specific configurable parameters. Selections are made by clicking on the appropriate radio button or pull-down menu option. The Terminal-Endpoint ID can be adjusted, within the allowable range, using the arrow keys on your keyboard or by clicking on the appropriate arrow image.

T-1 Carrier **E-1 Carrier**

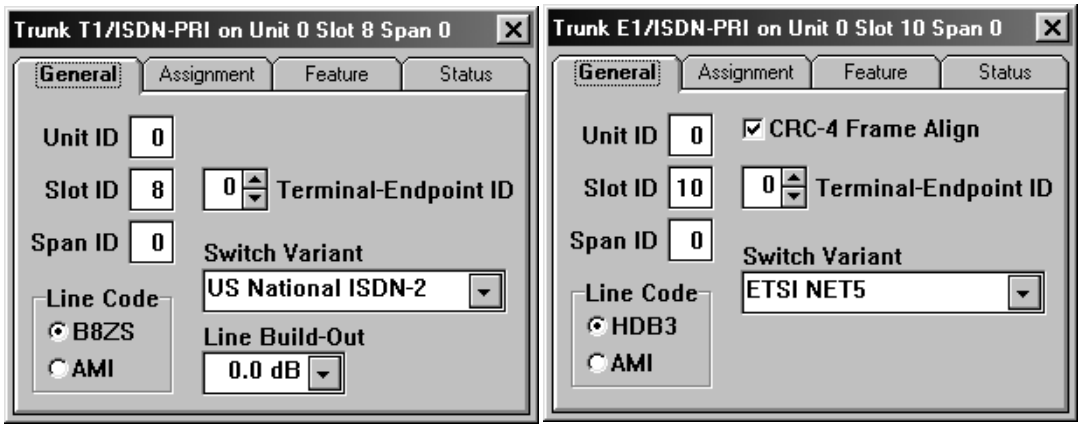


Table 11 - 1 ISDN PRI: General

Unit ID	**T1**	**E1**
Logical identifier for the system being configured. Unless there is more than one system, as in a linked environment, this ID will be 0.	Automatically Assigned	Automatically Assigned
Slot ID		
Logical identifier for the slot in which the card is inserted.	Automatically Assigned	Automatically Assigned
Span ID		
Logical identifier for the span being configured.	Automatically Assigned	Automatically Assigned
Line Code (Glossary page 26)		
Method of manipulating the bit pattern to maintain synchronization between the central office and the card.	B8ZS* AMI	HDB3* AMI
CRC-4 Frame Align		
Instructs the card to send cyclic redundancy check (checksum) information to the central office.		Disabled* Enabled

Table 11 - 1 ISDN PRI: General

Terminal-Endpoint ID (TEI) (Glossary page 27)		
Device identifier for the PRI.	0* Range: 0-63	0* Range: 0-63
Switch Variant		
The ISDN PRI protocol supported by the central office switch.	US National ISDN-2* AT&T 4ESS AT&T 5ESS Nortel DMS-100	ETSI NET5* Australia TS014.1
Line Build-Out (Glossary page 26)		
The amount of line build-out (LBO) necessary to attenuate signals transmitted from the card so that they are properly received and handled by receiving equipment.		0.0 dB* -7.5 dB -15.0 dB -22.5 dB

* = default settings

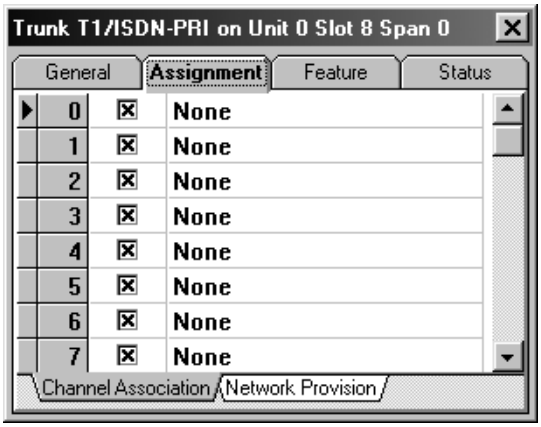
11.5.3 Assignment: Channel Association

The assignment: channel association tab includes checkboxes for enabling and disabling channels and pull-down menus for assigning trunks to channels.

Tip
To quickly assign a range of trunks to the enabled B-channels, right-click anywhere in the third column and select AutoAssign. To quickly enable / disable channels, right-click anywhere in the second column and choose Auto Assign to keep only assigned channels enabled, enable ALL channels, or disable ALL channels.

.

T-1 Carrier



E-1 Carrier

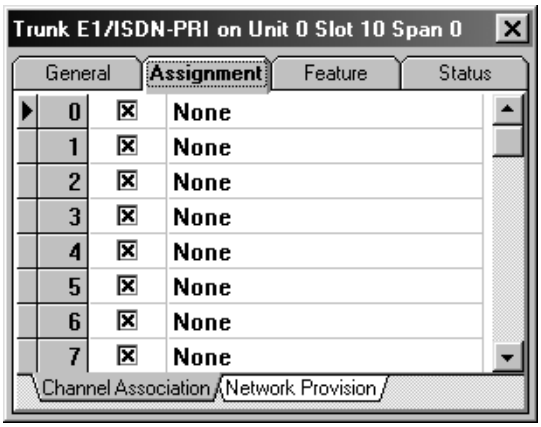


Table 11 - 2 ISDN PRI: Assignment: Channel Association

Channel ID	T-1	E-1
Logical IDs for available B-channels. These IDs are not meant to correspond to actual time slot IDs in the framing pattern.	0 - 22, 22*	0 - 29, 29*
Checkbox		
Enables/disables the corresponding B-channel.	Enabled* Disabled	Enabled* Disabled
Trunk Assignment (See "Channel Selection Order" and "Key Telephone Call Appearance" in the Glossary)		
The pull-down menu is used to logically assign a trunk to each channel. This assignment only establishes the direction in which incoming calls are dynamically assigned to trunks. There is no physical relationship between a trunk and a channel.	None* Trunk ID	None* Trunk ID

* = default settings

11.5.4 Assignment: Network Provision

The assignment: network provision tab provides an input box to enter the main directory number associated with your PRI and a pull-down menu that allows you to select the order in which channels are utilized for outbound calls. The directory number should be obtained from your provider when you activate service.

The main directory number is also referred to as "Billing Telephone Number (BTN)" and 'Telephone Line Identification (TLI)."

T-1 Carrier **E-1 Carrier**

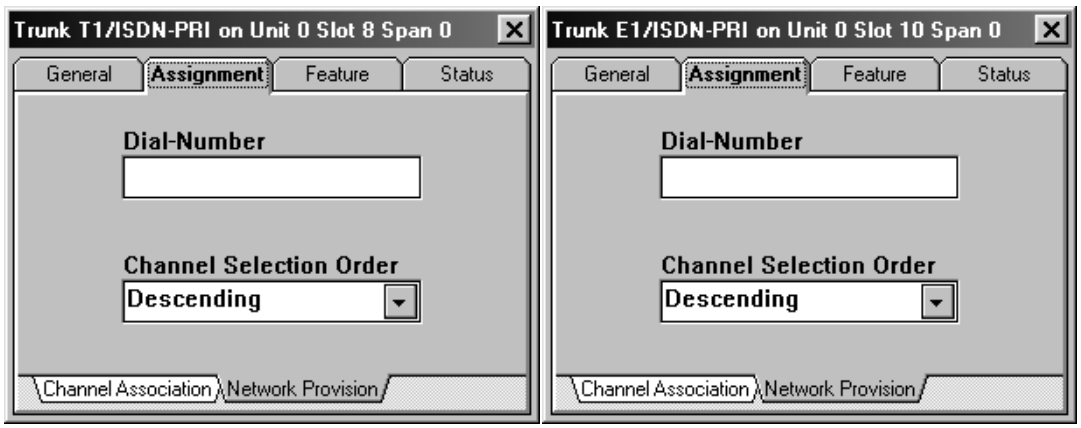


Table 11 - 3

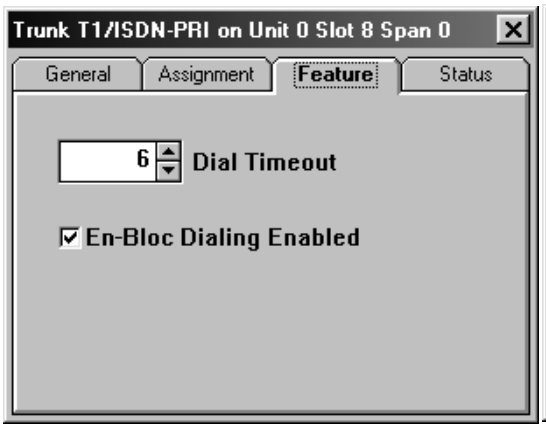
Dial-Number (Glossary page 22)	**T-1**	**E-1**
The main directory number for the PRI This number is presented to the network on outbound calls in the absence of an Outbound Caller-ID tag.	up to 20 digits	up to 20 digits
Channel Selection Order (Glossary page 22. See also "Key Telephone Call Appearance" in the Glossary)		
The direction in which the B-channels are used for outbound calls.	Descending* Ascending	Descending* Ascending

* = default settings

11.5.5 Feature

The feature tab includes a programmable timing parameter and a checkbox to enable/disable en-bloc dialing.

T-1 Carrier



E-1 Carrier

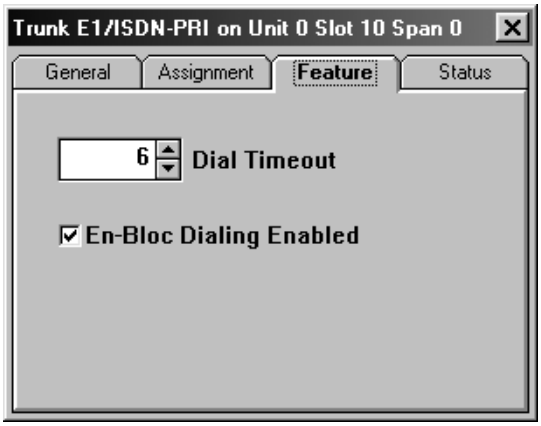


Table 11 - 4 ISDN PRI: Feature

Dial Timeout	T-1	E-1
The period of time that must elapse before the system assumes that the user is finished dialing.	6 sec* Range: 0-255 sec	6 sec* Range: 0-255 sec
En-Bloc Dialing Enabled (Glossary page 23)		
Enables/disables en-bloc dialing. En-bloc dialing is the recommended dialing method on an ISDN PRI.	Enabled* Disabled	Enabled* Disabled

* = default settings

11.5.6 Call Routing

In order to use Calling Number Delivery and Called Party Number information to route calls, you must first establish a relationship between each calling number/called party number and the user, user group, or other system entity to which you want the call routed. These relationships are defined on the system's route map. See System Parameters: Route Map in the Software Configuration section of the Plexus System Manual.

If you obtain a large enough block of directory numbers from the provider, a unique number can be assigned to each user on the system. Using the Called Party Number information received on the D-channel and a DNIS assignment on the system's route map, calls can then be directly routed to users based on the number dialed, as on a DID trunk. If the user's assigned directory number corresponds with their Outbound Caller-ID tag, return calls can be placed to the user based on the Caller ID.

Note
Check with your provider about getting additional directory numbers. Typically, directory numbers are available in blocks.

11.5.7 Outbound Caller-ID tag

An Outbound Caller-ID tag should be defined for each user that will be placing calls on the PRI [see User: Feature in the Software Configuration section of the Plexus System Manual]. If the Outbound Caller-ID tags correspond to legitimate directory numbers, the number is typically forwarded on outbound calls as the Caller ID[1]. If the Outbound Caller-ID tags are not directory numbers, they can still be used to track calls on a user-basis as many providers record the Outbound Caller ID data and present it on their billing statements.

1. Check with your provider.

11.6 Status Monitoring

Plexus Administrator provides an interface which can be used to monitor the real-time status of your installed and configured PRI.

To monitor your PRI, proceed as follows:

1. Launch Plexus Administrator.
2. Open the appropriate configuration file.
3. Select **Open** from the Link menu.
4. Click on either port on the image of the ISDN PRI card.
5. Click on the **Status** tab.

11.6.1 Status

The status tab includes a status indicator for each installed span.

T-1 Carrier **E-1 Carrier**

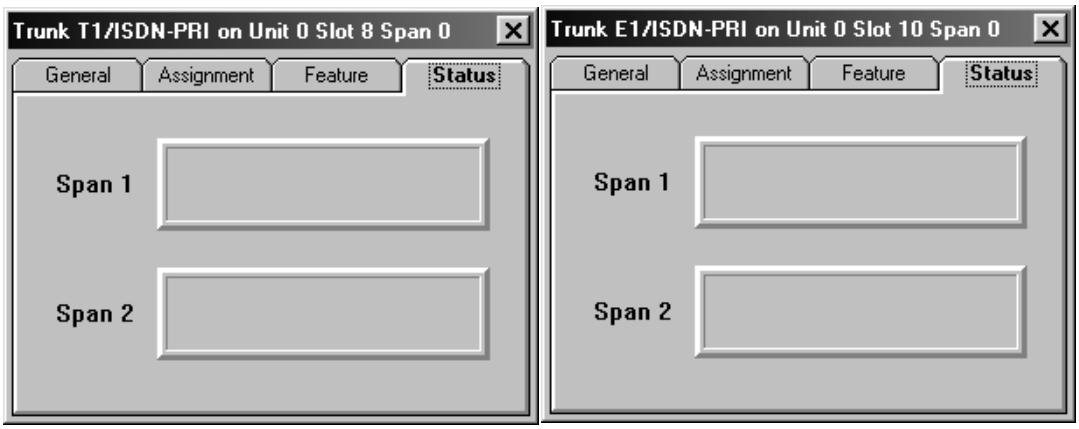


Table 11 - 5 ISDN PRI: Status

Span #	T-1 / E-1
Indicates the state of the signal on the corresponding span and the number of channels currently in use. Different colors indicate different states.	**Green** = good signal **Red** = no signal

11.7 Glossary

11.7.1 Called Party Number

Also known as DNIS; the number that the caller dialed.

11.7.2 Calling Number Delivery

Also known as ANI and Caller ID; the caller's phone number.

11.7.3 Channel Selection Order

The Channel Selection Order has to do with the direction the channels are used for outbound calls. Channels can be used in an ascending direction or in a descending direction. For each outbound call, the PRI card requests the next available channel in the selected direction. For example, on a T-1 carrier, if descending is selected and channels 22 and 23 are in use, the Plexus will request channel 21. Note that the central office can grant or deny the channel request.

To avoid glare, opposite directions should be selected for inbound and outbound calls. When you activate your PRI, your provider may allow you to choose the direction channels are selected for inbound calls. Inbound calls typically select channels in an ascending direction.

11.7.4 Clear

Indicates that only user data is on these channels; no bits are robbed or stuffed for signaling or framing.

11.7.5 Dial-Number

The Dial-Number is the main directory number associated with the PRI and the number presented to the network on outbound calls for Caller ID purposes. Calls made by users with an assigned Outbound Caller-ID tag that corresponds to a legitimate directory will present their tag instead of the Dial-Number.

The **main directory number** is also referred to as "Billing Telephone Number (BTN)" and 'Telephone Line Identification (TLI)."

11.7.6 DNIS

Dialed Number Identification Service; the number that the caller called.

11.7.7 En-Bloc vs. Overlap

En-bloc and overlap are methods of sending the dialed digits to the central office on an outbound call. The en-bloc method sends the complete number in a single setup message. Dialed numbers are essentially buffered until dialing is complete; similar to dialing a number on a cell phone and then pressing send. The overlap method sends the number in separate messages, one digit at a time. En-bloc is the universally required method on an ISDN PRI. Overlap on PRI will only work in special, customized applications.

Note
Users can indicate that they are done dialing by pressing #. The system will immediately send the en-bloc message (i.e., the dialed number) when it detects the #, otherwise the message is not sent until the dial timeout expires.

11.7.8 Fractional T-1/E-1

A span with less than the full number of channels enabled.

11.7.9 Glare

Collisions resulting from an attempt to seize a channel on which an inbound call is being received.

11.7.10 HDB-3 and B8ZS compared

The primary difference between HDB-3 used on E-1 and B8ZS used on T-1 are the number of consecutive zeros they allow. HDB-3 replaces 4 consecutive zeros with a bipolar violation (successive pulses with the same polarity). B8ZS replaces a string of 8 zeros with a pre-defined bit pattern.

11.7.11 ISDN

Integrated Services Digital Network

11.7.12 Key Telephone Call Appearance

The following configuration is recommended in order to maximize call appearance on key telephones connected to the system.

- Select **Descending** for the Channel Selection Order.
- Create a single trunk group for all of the trunks assigned to the PRI.

- Assign the trunks to the trunk group in ascending order.
- Request that your provider implement ascending channel selection for inbound calls.

This configuration causes both inbound and outbound calls to be assigned to the trunks in an ascending direction. Each new call is assigned to the first available trunk in ascending order. Therefore, even with a limited number of DTS keys, the majority of calls will have an associated call appearance.

The following narratives and diagrams illustrate the benefits of the recommended configuration.

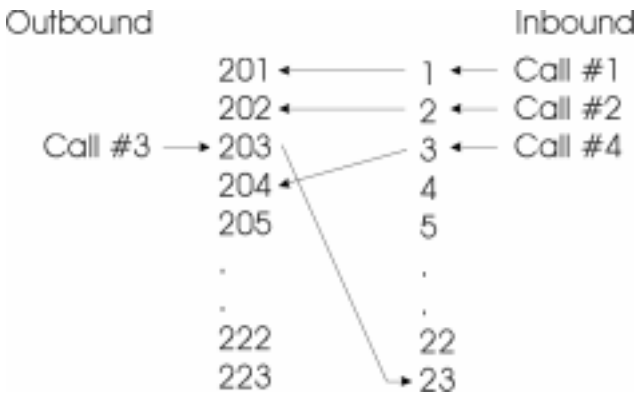


Table 11 - 6 Frame 1

1 Inbound call comes in on channel 1 and is assigned to trunk 201.

2 Inbound call comes in on channel 2 and is assigned to trunk 202.

3 Outbound call is placed. Trunk 203 is assigned to the call and the Plexus system requests and ultimately seizes channel 23.

4 Inbound call comes in on channel 3 and is assigned to trunk 204. Trunk 204 is the first available trunk, not necessarily the trunk that we assigned to the channel.

Note
There is no physical relationship between the trunks and the channels (see “Trunk Assignment page 17).

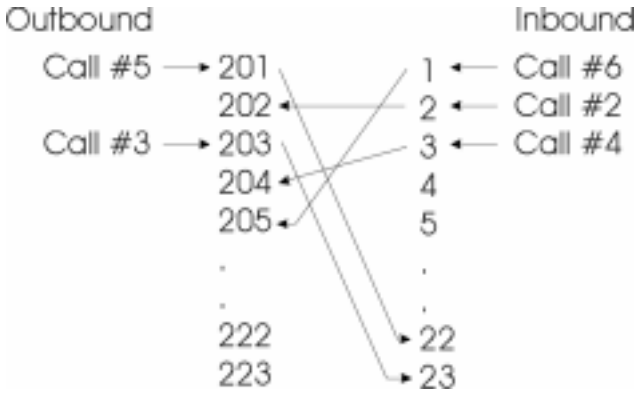


Table 11 - 7 Frame 2

5 Call #1 completes.

6 Outbound call is placed. Trunk 201 (the first available trunk) is assigned to the call and the Plexus system requests and ultimately seizes channel 22 (the first available channel in the descending direction).

7 Inbound call comes in on channel 1 and is assigned to trunk 205. Once again, trunk 205 is the first available trunk, not necessarily the trunk that we assigned to the channel.

11.7.13 Line Build-Out

A Line Build-Out (LBO) attenuates the signal transmitted by the PRI card (T-1 carrier only). LBO simulates cable loss to ensure that network equipment can properly receive and handle the transmitted signal. There are various equations used to determine the LBO value that is needed to properly attenuate a signal. Fortunately, your provider's installation personnel are responsible for informing you as to which LBO code applies at your installation. This information is used to determine the option you select in Plexus Administrator, as follows:

Code	Setting	w/ additional cable loss of 7.5 - 13.0 dB	w/ additional cable loss of 15.0 - 20.5 dB
A	0.0 dB	0.0 dB	0.0 dB
B	-7.5 dB	0.0 dB	0.0 dB
C	-15.0 dB	-7.5 dB	0.0 dB

Additional cable loss is achieved by moving the Plexus system further from the demarcation point. Unless the system needs to be relocated, it is recommended that you go with the advised code and retain the current position of the system. Otherwise, you will need to measure cable loss and determine whether you need to modify the LBO setting.

Note
The -22.5 dB setting is typically not used and is reserved for special applications.

11.7.14 Line Code

The line code has to do with how the binary data is coded. The line code options were developed in response to different challenges identified in receiving and synchronizing on the data bit pattern. AMI (alternate mark inversion) manipulates the data by reversing the polarity of successive 1's or marks. This helps the receiving equipment to maintain a zero voltage reference point.

B8ZS (binary 8-zeros suppression) and HDB3 (High Density Bipolar 3-zeroes), both developed subsequent to AMI, deal with “ones density” - the frequency of 1’s in the data bit pattern. Ones density is important because PRI devices synchronize by tracking the 1’s. A long string of 0’s increases the likelihood that a device will be unable to maintain synchronization. While today’s equipment is better able to deal with a long string of 0’s, B8ZS and HDB3 are still viewed as improved line code alternatives and are therefore recommended.

The most important consideration when selecting the line code option is that the same option be used on both ends. Therefore, make sure that you and the provider agree on the selected option.

11.7.15 POTs:

Plain Old Telephone lines.

11.7.16 PRI

Primary Rate Interface

11.7.17 Span

A group of channels or time slots.

11.7.18 Terminal-Endpoint ID

The Terminal-Endpoint ID (TEI) is a logical identifier that must be set for each device connected to an ISDN line. The central office uses the ID to identify the different devices so that it can maintain the necessary number of logical links on the D-channel.

The Plexus ISDN PRI card is "Nonautomatic TEI assignment user equipment" which means that a TEI must be manually programmed. TEIs 0-63 are allocated to this class of equipment.

Since PRIs are typically used as trunks connected to a single PBX, a ‘0’ setting is usually acceptable. Note, however, that a manually programmed TEI must be verified and accepted by the central office prior to its use. Therefore, you should discuss the TEI value with your provider when you activate service.

Note
TEI settings are more important in the case of BRI, where up to 8 devices can be connected to a single ISDN. However, most BRI equipment is "Automatic TEI assignment user equipment," which means it is capable of requesting an automatic TEI assignment from the central office.

ISDN BRI

12

12.1 Overview

ISDN BRI is a communication protocol that utilizes existing local loop wiring (i.e., standard 2-wire, single pair) to carry a digital signal. The BRI allocates the full bandwidth of two 64 Kbps channels, called B-channels, to the transmission of voice and/or data. All network signaling is carried on a third channel, the D-channel, using a standard packet protocol. The D-channel on a BRI operates at 16Kbps.

This chapter covers the installation, configuration, and troubleshooting of the ISDN BRI peripheral card for the Plexus Hybrid Digital Telephone System. The chapter includes the following sections:

Hardware Description	**A description of the hardware, including the connectors and LEDs, and the specifications.**
Installation	Instructions on installing the ISDN BRI card and connecting it to the network interface.
Configuration	Instructions on using the real-time diagnostics available in Plexus Administrator and interpreting the status indicators.
Troubleshooting	Instructions on interpreting the real-time status indicators available in Plexus Administrator.

12.2 Hardware Description

The ISDN BRI peripheral card supports up to four (4) BRIs. There are versions of the card for both the U and the S/T interface. The difference between the two interfaces has to do with the network boundary. In the case of the S/T interface, the network boundary extends to the NT1 equipment and the provider is therefore responsible for maintaining it, thus reducing the cost of S/T terminal equipment. The network boundary, in the case of the U interface, does not extend to the NT1 which must be built in to the terminal equipment.

The network boundary is the point at which the public network ends and the customer premises begins.

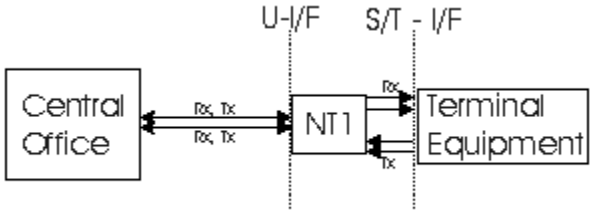


Table 12 - 1 Network Boundary - U vs S/T

The NT1 is responsible for line performance monitoring, timing, physical signaling protocol conversion, electrical conversion, and power transfer. The NT1 converts the single pair from the central office to a two-pair (4-wire) connection; 2 wires for transmit and 2 for receive. The U interface version of the card includes the NT1 equipment on the card. The U interface is standard in the US. The S/T interface is the standard almost everywhere else.

Note
The S/T version of the ISDN BRI card does not support ISDN point-to-multipoint applications. However, because the Plexus is a switch, multiple devices can utilize the BRI. The BRI is essentially a trunk on the system.

12.2.1 Connectors

The ISDN BRI card includes four 8-pin connectors that accept either RJ-11 or RJ-45 modular plugs. Note that the pin out differs depending on the interface version.

U Interface Pin Out	
Pin	**Signal**
4	Tx, Rx
5	Tx, Rx

S/T Interface Pin Out	
Pin	**Signal**
3	Tx Ring
4	Rx Ring
5	Rx Tip
6	Tx Tip

12.2.2 LEDs

A red LED is provided for each BRI. Two different states provide status information as follows:

LED states	
On	One or both B-channels are in use
Off	Not in use

12.2.3 Specifications

U interface	S/T interface
ANSI T1.601	ITU-T I.430

12.3 Installation

ISDN BRI installation and activation are handled by the service provider. In most markets there are several providers to choose from. The provider you select should be based on several factors, including rates.

When you place your order, make sure that you get the following information from the provider.

- Two (2) main telephone numbers
- Additional numbers, if you plan to order any
- SPIDs
- Dialing method - en-bloc or overlap
- Supported standard (e.g., National ISDN-1, ETSI NET3)
- Confirmation of supplemental services ordered (e.g., Calling Number Delivery, also known as Caller ID)

Once the order is placed, the provider will dispatch technicians to drop a cable into your place of business. The cable will likely be terminated into a biscuit jack with an 8-pin connector. If you are using a U interface (in the US), the jack will only use the two innermost pins. If an S/T interface is installed, the jack will use the four innermost pins (see “Connectors” on page 3 for pin out information). This jack is also known as the demarcation point or network interface. You are responsible for the connection between the ISDN BRI card and this jack.

Note
Because there are so few configurable parameters with an ISDN BRI, the interface will likely be active upon completion of the installation.

12.3.1 Inserting the card

1. Power off the system.
2. Carefully remove the card from the packaging.

Note
The cards are sensitive to static electricity and should be handled by the edges only. Never touch the components on a card.

3 Place the edges of the card inside the top and bottom card guides of any slot except for 9 and 10.

4 Gently slide the card in until it completely seats into the backplane. There will likely be an increase in pressure and a slight noise as the card is seated.

12.3.2 Making the connection

Using a cable that conforms to the following wiring diagram, run cable between the connector(s) on the ISDN BRI card and the demarcation point(s). Note that the wiring differs depending on the interface version

U Interface	
Straight Through	
4	4
5	5

S/T Interface	
Straight Through	
3	3
4	4
5	5
6	6

The innermost pins are used in both cases, 2 pins for the U interface and 4 for S/T interface. Either an RJ-11 or an RJ-45 modular plug can be used.

Tip
Since the wiring is straight through, a standard 10 BASE T cable can be used for the above connection.

1 Insert one end of the cable into one of the connectors on the card.

2 Insert the other end into the biscuit jack installed by the provider.

Note
If you are installing fewer than 4 BRIs, use the connectors from the top down.

12.4 Configuration

Configuration of the ISDN BRI card consists of entering the telephone numbers and SPIDs, selecting the supported standard, and defining the trunks that will make use of the BRI.

Configuration is typically performed after the card is inserted and connected to the demarcation point. Each parameter on each tab should be addressed in the order that they are presented. Below, you will find an explanation of each parameter, the available options, and an indication of the default setting.

12.4.1 General

The general tab includes ID information and radio buttons that allow you to select the ISDN standard supported by your central office switch. Some switches in the US do not support National ISDN-1. The ETSI NET3 standard is used for S/T interfaces throughout Europe.

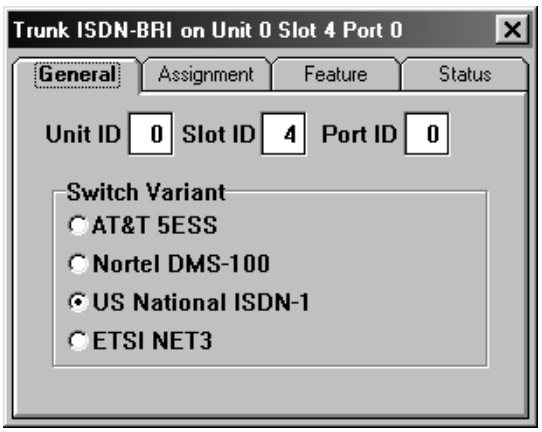


Table 12 - 2 ISDN BRI: General

Unit ID	
Logical identifier for the system cabinet being configured. ***Note:*** Unless there is more than one system, as in a linked environment, this ID will be 0.	Automatically Assigned

Table 12 - 2 ISDN BRI: General

Slot ID	
Logical identifier for the slot in which the ISDN BRI card is inserted.	Automatically Assigned
Port ID	
Logical identifier for the BRI being configured. ***Note:*** The ISDN BRI card supports up to 4 BRIs.	Automatically Assigned
Switch Variant	
The ISDN BRI protocol supported by the central office switch. Some US switches may not support the national standard.	AT&T 5ESS Nortel DMS-100 US National ISDN-1* ETSI NET3

* = default settings

12.4.2 Assignment: Network Provision

The assignment: network provision tab provides input boxes for the main ISDN phone numbers and SPID numbers associated with the BRI. This information should be obtained from your provider either when you place the order or when the cable is dropped at the customer premises.

Note
Additional directory numbers can be obtained from your provider. Different providers impose different limits on the number of directory numbers they will assign for a BRI.

Note
SPIDs are not required for Switch Varient ETSI NET3.

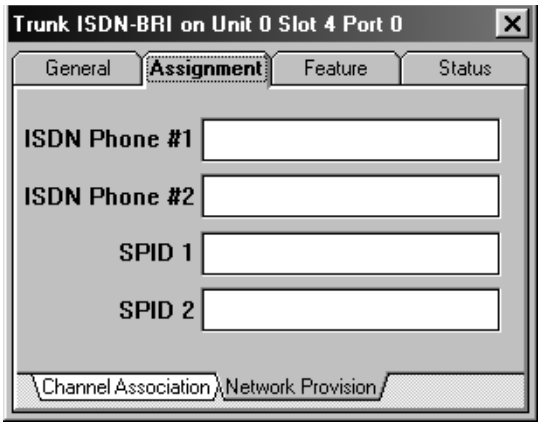


Table 12 - 3 ISDN BRI: Assignment: Network Provision

ISDN Phone #1 and #2	
The main directory numbers assigned to the BRI. These numbers are presented to the network on outbound calls in the absence of an Outbound Caller-ID tag. See “Outbound Caller-ID tag” on page 13.	up to 20 digits
SPID 1 and 2	
Service Profile Identifier (SPID). SPIDs are assigned by the provider. They are used by the network to determine the class of service that corresponds to the terminal equipment. ***NOTE:*** The ISDN BRI Interface card will not operate if the SPIDs are not configured correctly.	Up to 20 digits

12.4.3 Assignment: Channel Association

The assignment: channel association tab includes checkboxes for enabling and disabling channels and pull-down menus for assigning trunks to channels.

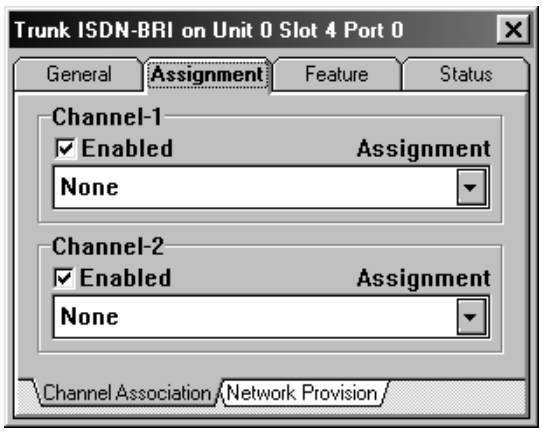


Table 12 - 4 ISDN BRI: Assignment: Channel Association

Channel-# Enabled	
Enables the corresponding B-channel. This box should always be checked unless you have a non-standard application (e.g., 1 B+D),	Enabled (checked)* Disabled (unchecked)
Channel-# Assignment	
The pull-down menu is used to logically assign a trunk to each channel. This assignment only establishes the trunks that will utilize the BRI. There is no physical relationship between a trunk and a channel.	None* Trunk ID

* = default settings

Tip
To quickly assign trunks to the enabled B-channels, right-click anywhere in the assignment input box and select AutoAssign.

12.4.4 Feature

The feature tab includes a programmable timing parameter and a checkbox to enable/disable en-bloc dialing.

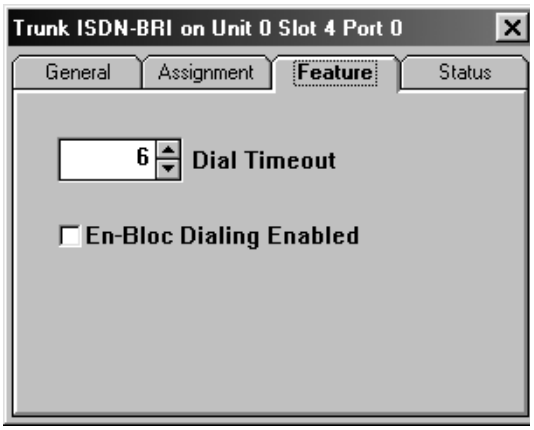


Table 12 - 5 ISDN BRI: Feature

Dial Timeout	
The period of time that must elapse before the system assumes that the user is finished dialing. The Dial Timeout is an interdigit timeout that resets each time a digit is dialed before the period expires. If the timeout expires before another digit is dialed, the system assumes that the user is done dialing and regenerates the dial string.	6 seconds* 0-255 seconds
En-Bloc Dialing Enabled	
Enables/disables en-bloc dialing. En-bloc dialing sends the dialed number as a single message on the D-channel. The alternative method, called overlap, sends the dialed number in separate messages, one digit at a time. Overlap (i.e., en-bloc not enabled) is the recommended method on a BRI.	No (unchecked)* Yes (checked)

* = default settings

Note
En-Bloc vs. Overlap: En-bloc and overlap are methods of sending the dialed digits to the central office on an outbound call. The en-bloc method sends the complete number in a single setup message. Dialed numbers are essentially buffered until dialing is complete; similar to dialing a number on a cell phone and then pressing send. The overlap method sends the number in separate messages, one digit at a time. En-bloc is the universally required method on an ISDN PRI. Overlap on PRI will only work in special, customized applications.

12.4.5 Automatic Call Routing

Calling Number Delivery (Caller ID) and Called Party Number (CPN) information received on your ISDN BRI can be used to automatically route calls to specific users or user groups on the system, bypassing the auto-attendant. For example, you can route all callers that dial one (or more) of your assigned directory numbers to the sales group and those that dial one (or more) of the other numbers to the tech support group.

In order to use Calling Number Delivery and Called Party Number to route calls, you must establish a relationship between each calling number/called party number and the user, user group, or other system entity to which you want the call routed. These relationships are defined on the system's route map[1].

12.4.6 Outbound Caller-ID tag

Outbound calls made on a BRI can be tagged by the Plexus system with an ID known as the Outbound Caller-ID Tag. Each user on the system can have their own ID. If the IDs correspond to legitimate directory numbers (a directory maintained by the service provider) for the BRI they are sent as the Caller ID information. If a relationship is defined between the directory number and the user on the system's route map[1], the Outbound Caller-ID Tag can also serve as the user's DID number.

The Outbound Caller-ID tag can also be used for call reporting. Most providers record the outbound Caller ID data and present it on their billing statements. This allows customers to track their expenses on a user-basis.

1. See System Parameters: Route Map in chapter 6, “Software Configuration” of this Manual.

12.5 Troubleshooting

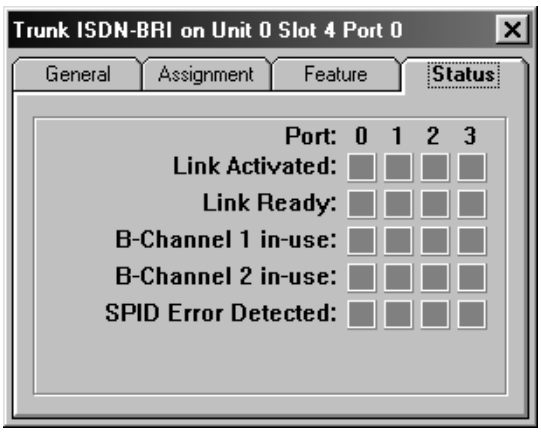
Plexus Administrator provides an interface which can be used to monitor and troubleshoot your installed BRI.

To monitor your BRI, proceed as follows:

1. Launch Plexus Administrator.
2. Open the appropriate configuration file.
3. Select **Open** from the Link menu.
4. Click on one of the ports on the image of the ISDN BRI card.
5. Click on the **Status** tab.

12.5.1 Status

The status tab includes several indicators which provide real-time diagnostic information. The various indicators provide information about the state of both the received and the transmitted signal.



Port	
Identifies the BRI to which the status indicators correspond. There is a column of indicators for each of the 4 possible BRI. The ports are numbered 0-3 from the top down.	0 1 2 3

13

Direct Inward Dialing (DID)

13.1 Overview

Direct Inward Dialing (DID) is a service offered by telephone companies in which the last digits (typically two to four) dialed by the caller are forwarded to the Plexus system on a DID trunk. Calls are routed within the Plexus system based on these digits.

DID allows the system to have several inbound dialed numbers that route directly to different users, user groups, auto attendants, etc. DID also allows the system to have more telephone numbers than it has trunks because not all telephone numbers will be in use at the same time.

Note
DID trunks can only be used for incoming calls. Users are not able to make outgoing calls using these trunks.

The Plexus system offers two types of DID:

13.1.1 DID

DID uses the digits sent by the telephone company on a DID trunk to route the calls directly. The calls are routed to the extension matching the DID digits. E.g., if the telephone company sent the digits 101 on the DID trunk, the call would automatically route to extension 101.

To configure DID, see “DID Configuration” - section 13.5.

13.1.2 Mapped DID

Mapped DID uses the digits sent by the telephone company on a DID trunk to route the calls based on information entered into a table. This allows for the DID digits to be any set of numbers, not exactly matching the system extension numbers. For an example, see “Mapped DID” - section 13.3.

To configure Mapped DID, see “Mapped DID Configuration” - section 13.6.

13.2 DID

For Direct Inward Dialing (DID) to work properly, the digits forwarded to the Plexus system by the phone company must match the IDs of system entities. A DID call can be routed to users, user groups, trunks, trunk groups, or auto attendants.

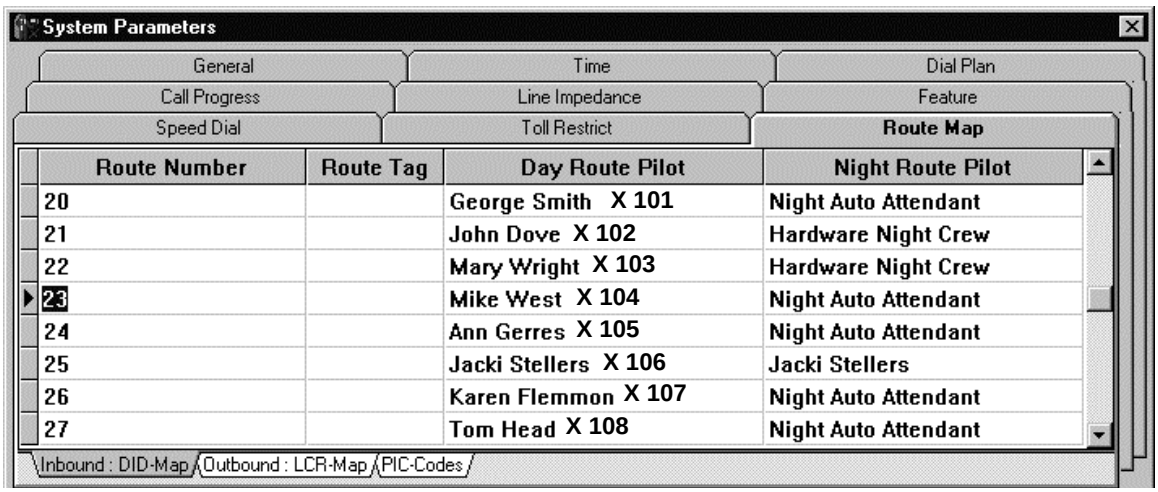
When the system receives the digits form the telephone company on a DID trunk, the system immediately forwards the call to the system entity ID that matches the sent digits. If the numbers do not match any system entity, the call will fail.

Note
Ensure that the numbers being sent by the telephone company to a DID trunk with DID routing match the IDs of system entities. If digits are received that do not match any system entity ID, the call will fail.

13.3 Mapped DID

13.3.1 Example

All the phone numbers from 555-0000 to 555-0049 (**50** distinct telephone numbers) could be assigned to a customer with **8** DID trunks. When a caller dials any number in this range, the call is forwarded by the telephone company to any of the available trunks. In the following example, if the caller dialed 555-0023, then the digits 2 and 3 will be forwarded. The system will read the digits and route the call to the target user based on the DID route map.



In this example, if the caller dialed 555-0023, the call would be routed directly to Mike West at extension 104 during the day and to the Night Auto Attendant at night.

13.4 Installation

13.4.1 Insert the DID Peripheral Card

The DID card is inserted into the system cabinet like any other peripheral card. Refer to the chapter 4, "Installation" of this manual for complete information on peripheral card installation.

13.5 DID Configuration

For DID to work properly, several parameters must be configured. The parameters, working together, will route the incoming call to the proper target entity. The following windows are where the DID parameters are set:

- Trunk: Routing: Day/Night: Voice: Routing Method
- Trunk Port: General,
- Trunk Port: Features,
- Trunk Port: MFC Signaling, and

13.5.1 Trunk: Routing: Day/Night: Voice: Routing Method

Table 13 - 1 Trunk: Routing: Voice: Day / Night Calls

Route-Method	
Method by which the system will route calls.	Fixed * DID Mapped-DID / DNIS Caller-ID / ANI

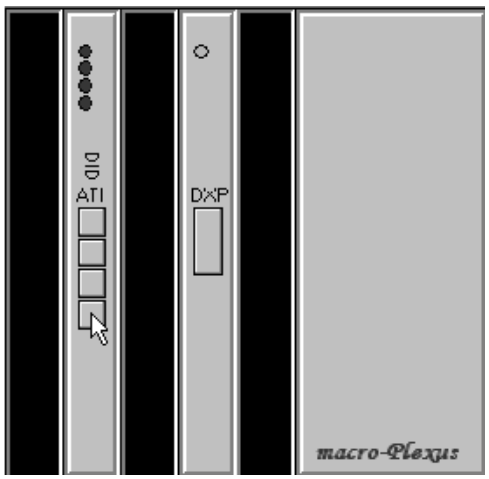
Table 13 - 1 Trunk: Routing: Voice: Day / Night Calls

Pilot	
Not applicable. All calls will be routed directly to the system entity ID that matches the DID digits received. Any calls with DID digits not matching a system entity ID will fail.	None * User ID User Group ID AutoAttendant ID Voice ID

* = default settings

13.5.2 Trunk Port: General

The Trunk Port windows are accessed by left clicking on any port of the DID card:



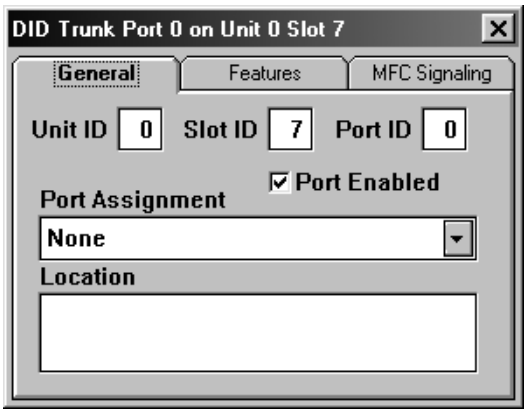


Table 13 - 2 Trunk Port: General

Unit ID	
Logical Identifier for the unit (i.e., system cabinet) being configured. ***Note:*** Unless there is more than one system (i.e., a linked environment), this ID will be 0.	Automatically Assigned
Slot ID	
Logical Identifier for the slot which the Direct Inward Dial (DID) Interface card is inserted.	Automatically Assigned
Port ID	
Logical Identifier for the trunk port being configured.	Automatically Assigned
Port Enabled	
Enables the trunk port	Enable* /Disable
Port Assignment	
The assigned trunk **TIP:** Right mouse-click in the box to utilize the Auto Assign feature.	Trunk ID/None*
Location	
Informal description or notes about the trunk.	Text box

* = default settings

13.5.3 Trunk Ports: Features

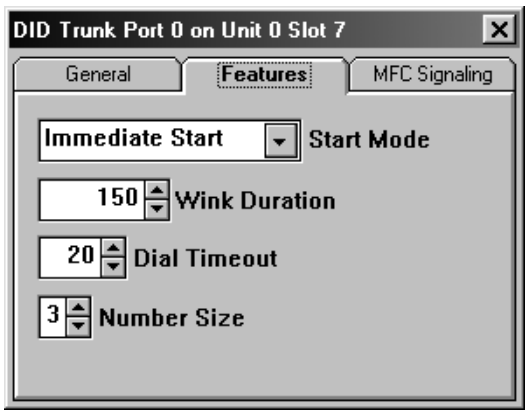


Table 13 - 3 Trunk Port: Features

Start Mode	
The method that the telephone company switch and the DID trunk port link.	Immediate Start* / Delay Start / Wink Start
Wink Duration	
Duration of Wink Start signal sent by the system to the telephone company switch.	0 – 65535 milliseconds / 150*
Dial Timeout	
The length of time that the system will wait for completion of dialed number information. If all the information does not reach the system before the Dial Timeout, the system will route the call based on the trunk's default routing.	1 – 255 seconds / 20*
Number size	
The number of digits reserved for the DID portion of the phone number. This will be the last x digits of the caller's dialed number that are sent by the telephone company to the system. The Inbound DID Route Map will use these digits to route the call to the appropriate extension	1 – 8 digits / 3*

* = default settings

13.5.4 Trunk Ports: MFC Signaling: Received (Rx) MFC Codes

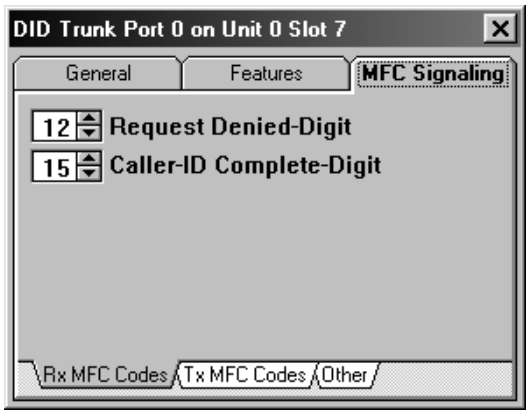


Table 13 - 4 Trunk Port: MFC Signaling: Rx MFC Codes

Request Denied-Digit	
Digit sent by the telephone company to indicate that the last request made by the system can not or will not be granted.	0 – 15 / 12*
Caller-ID Complete-Digit	
Digit sent by telephone company to indicate that the final character of the Caller ID information has been sent	0 – 15 / 15*

* = default settings

13.5.5 Trunk Ports: MFC Signaling: Transmitted (Tx) MFC Codes

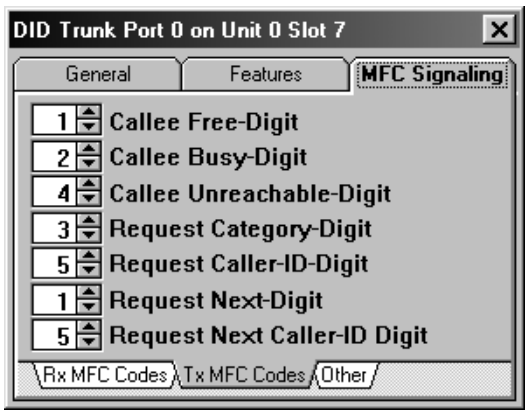


Table 13 - 5 Trunk Port: MFC Signaling: Tx MFC Codes

Callee Free-Digit	
Tells the telephone company that the target user is available and to send a ring-back tone to the caller.	0 – 15 / 1*
Callee Busy-Digit	
Tells the telephone company that the target user is busy / unavailable and to send a busy tone to the caller. ***Note:*** This signal will only be sent if the target user has no backup or coverage defined or if those entities are also unavailable.	0 – 15 / 2*
Callee Unreachable Digit	
Tells the telephone company that no entity is available to route the call to.	0 – 15 / 4*
Request Category-Digit	
Requests the category of the incoming call. This will vary by country but can include several categories, including: collect call, pay phone, residential phone, business phone, etc.	0 – 15 / 3*

Table 13 - 5 Trunk Port: MFC Signaling: Tx MFC Codes

Request Caller-ID-Digit	
Tells the telephone company to start sending the first Caller ID digit.	0 – 15 / 5*
Request Next-Digit	
After each digit of the called number is sent to the system, this signal states that the digit was received and asks for the next digit to be sent. After the last digit that the system can process has been received, this signal will not be sent.	0 – 15 / 1*
Request Next Caller ID Digit	
Tells the telephone company to start sending the next Caller ID digit.	0 – 15 / 5*

* = default settings

13.5.6 Trunk Ports: MFC Signaling: Other

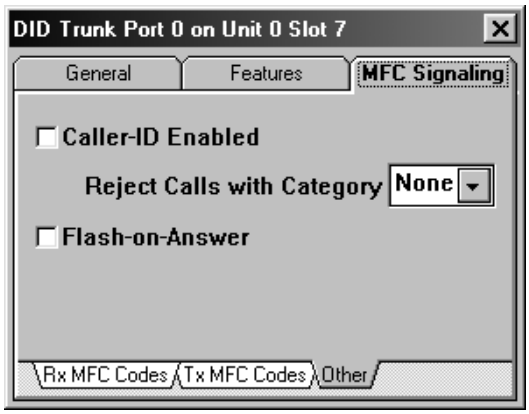


Table 13 - 6 Trunk Port: MFC Signaling: Other

Caller ID Enabled	
When enabled, the system will detect the Caller ID signal for each incoming call on the trunk. ***Note:*** Caller ID service must be activated on each incoming trunk through the telephone company in order for a Caller ID signal to be present.	Enable / Disable*

Table 13 - 6 Trunk Port: MFC Signaling: Other

Reject Calls with Category	
Rejects calls that receive this digit as the category of call. This digit is requested by the system from the telephone company. See Request Category-Digit in “Trunk Ports: MFC Signaling: Transmitted (Tx) MFC Codes” - section 13.6.5.	None*, 1-15
Flash on Answer	
Used to ’flash’ the CO on seizing an inbound call. This is used to signal the CO to reject certain types of calls (e.g., collect calls).	Enable / Disable*

* = default settings

13.6 Mapped DID Configuration

In order for Mapped DID to work properly, several parameters must be configured. The parameters, working together, will route the incoming call to the proper target entity. The following windows are where the DID parameters are set:

- Trunk: Routing: Day/Night: Voice: Routing Method
- Trunk Port: General,
- Trunk Port: Features,
- Trunk Port: MFC Signaling, and
- System Parameters: Route Map: Inbound: DID-Map

13.6.1 Trunk: Routing: Day/Night: Voice: Routing Method

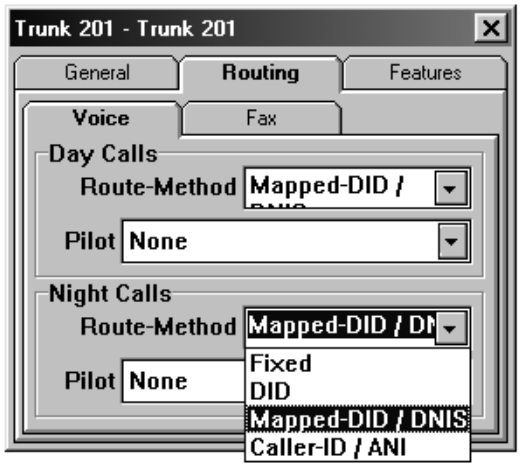


Table 13 - 7 Trunk: Routing: Voice: Day / Night Calls

Route-Method	
Method by which the system will route calls. ***Note:*** The Route Call during Day/Night should still be set if Fixed is not chosen. This will serve as the default routing in case other routing methods do not receive the required inputs from the local service provider (see “Trunk Routing” - section 8.133).	Fixed * DID Mapped-DID / DNIS Caller-ID / ANI

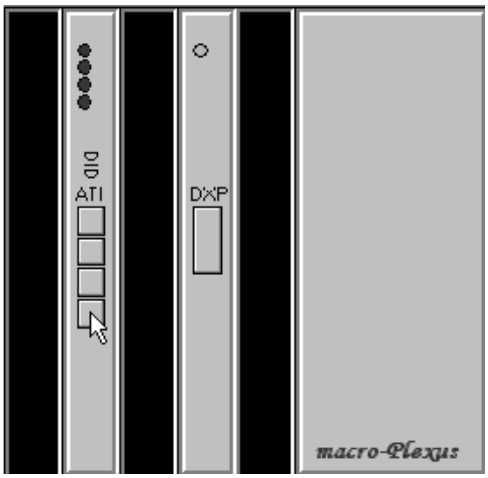
Table 13 - 7 Trunk: Routing: Voice: Day / Night Calls

Pilot	
Where the system will route the incoming calls during the day hours [see “System parameters: Time, Mode” - section 6.21.6]. ***NOTE:*** If any setting other than Fixed is set for Routing Method, this field will be used as the default routing in case other routing methods do not receive the required inputs from the local service provider.	None * User ID User Group ID AutoAttendant ID Voice ID

* = default settings

13.6.2 Trunk Port: General

The Trunk Port windows are accessed by left clicking on any port of the DID card:



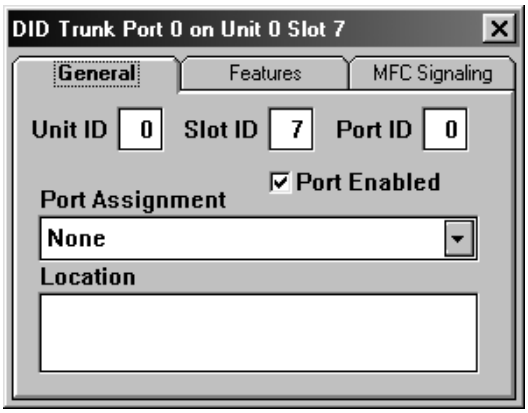


Table 13 - 8 Trunk Port: General

Unit ID	
Logical Identifier for the unit (i.e., system cabinet) being configured. ***Note:*** Unless there is more than one system (i.e., a linked environment), this ID will be 0.	Automatically Assigned
Slot ID	
Logical Identifier for the slot which the Direct Inward Dial (DID) Interface card is inserted.	Automatically Assigned
Port ID	
Logical Identifier for the trunk port being configured.	Automatically Assigned
Port Enabled	
Enables the trunk port	Enable* /Disable
Port Assignment	
The assigned trunk **TIP:** Right mouse-click in the box to utilize the Auto Assign feature.	Trunk ID/None*
Location	
Informal description or notes about the trunk.	Text box

* = default settings

13.6.3 Trunk Ports: Features

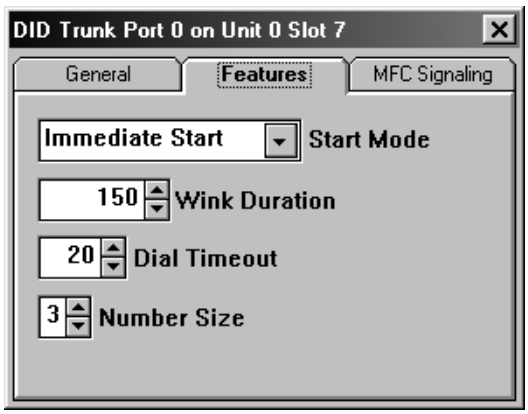


Table 13 - 9 Trunk Port: Features

Start Mode	
The method that the telephone company switch and the DID trunk port link.	Immediate Start* / Delay Start / Wink Start
Wink Duration	
Duration of Wink Start signal sent by the system to the telephone company switch.	0 – 65535 milliseconds / 150*
Dial Timeout	
The length of time that the system will wait for completion of dialed number information. If all the information does not reach the system before the Dial Timeout, the system will route the call based on the trunk's default routing.	1 – 255 seconds / 20*
Number size	
The number of digits reserved for the DID portion of the phone number. This will be the last x digits of the caller's dialed number that are sent by the telephone company to the system. The Inbound DID Route Map will use these digits to route the call to the appropriate extension	1 – 8 digits / 3*

* = default settings

13.6.4 Trunk Ports: MFC Signaling: Received (Rx) MFC Codes

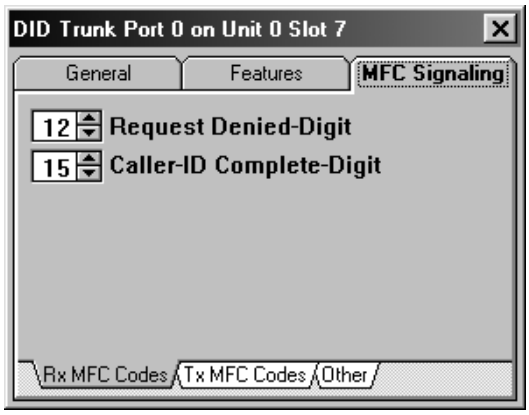


Table 13 - 10 Trunk Port: MFC Signaling: Rx MFC Codes

Request Denied-Digit	
Digit sent by the telephone company to indicate that the last request made by the system can not or will not be granted.	0 – 15 / 12*
Caller-ID Complete-Digit	
Digit sent by telephone company to indicate that the final character of the Caller ID information has been sent.	0 – 15 / 15*

* = default settings

13.6.5 Trunk Ports: MFC Signaling: Transmitted (Tx) MFC Codes

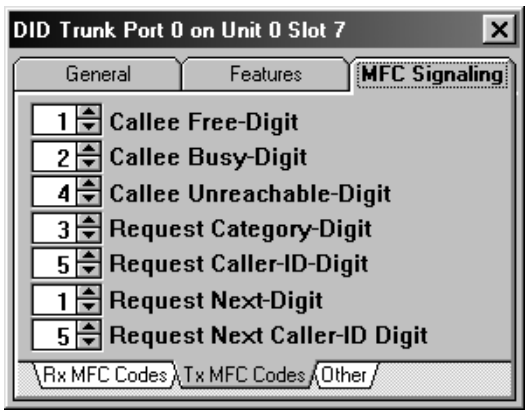


Table 13 - 11 Trunk Port: MFC Signaling: Tx MFC Codes

Callee Free-Digit	
Tells the telephone company that the target user is available and to send a ring-back tone to the caller.	0 – 15 / 1*
Callee Busy-Digit	
Tells the telephone company that the target user is busy / unavailable and to send a busy tone to the caller. ***Note:*** This signal will only be sent if the target user has no backup or coverage defined or if those entities are also unavailable.	0 – 15 / 2*
Callee Unreachable Digit	
Tells the telephone company that no entity is available to which to route the call.	0 – 15 / 4*
Request Category-Digit	
Requests the category of the incoming call. This will vary by country but can include several categories, including: collect call, pay phone, residential phone, business phone, etc.	0 – 15 / 3*
Request Caller-ID-Digit	
Tells the telephone company to start sending the first Caller ID digit.	0 – 15 / 5*

Table 13 - 11 Trunk Port: MFC Signaling: Tx MFC Codes

Request Next-Digit	
After each digit of the called number is sent to the system, this signal states that the digit was received and asks for the next digit to be sent. After the last digit that the system can process has been received, this signal will not be sent.	0 – 15 / 1*
Request Next Caller ID Digit	
Tells the telephone company to start sending the next Caller ID digit.	0 – 15 / 5*

* = default settings

13.6.6 Trunk Ports: MFC Signaling: Other

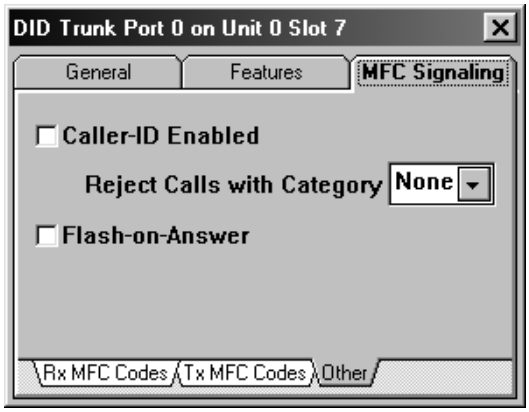


Table 13 - 12 Trunk Port: MFC Signaling: Other

Caller ID Enabled	
When enabled, the system will detect the Caller ID signal for each incoming call on the trunk. ***Note:*** Caller ID service must be activated on each incoming trunk through the telephone company in order for a Caller ID signal to be present.	Enable / Disable*

Table 13 - 12 Trunk Port: MFC Signaling: Other

Reject Calls with Category	
Rejects calls that receive this digit as the category of call. This digit is requested by the system from the telephone company. See Request Category-Digit in “Trunk Ports: MFC Signaling: Transmitted (Tx) MFC Codes” - section 13.6.5.	None*, 1-15
Flash on Answer	
Used to ’flash’ the CO on seizing an inbound call. This is used to signal the CO to reject certain types of calls (e.g., collect calls).	Enable / Disable*

* = default settings

13.6.7 System Parameters: Route Map: Inbound: DID-Map

The Inbound: DID-Map is used for multiple features. These include:

- Mapped DID / DNIS routing
- Caller ID / ANI routing (calls routed based on received Caller ID/ANI digits)

The type of routing for incoming calls is determined by the trunk routing (see “Trunk Routing” - section 8.133). The Inbound: DID-Map may contain several record entries used for the different routing methods. Each routing method will look for the information on the table that relates to its type of routing. For example, Mapped DID routing will only look at entries with the same number of digits that are defined as Number Size for DID routing. If the digits captured are 23 and 23 is listed on the Inbound DID map and 2355555 is listed in the table for Caller ID routing, the system will only look at the 23 entry.

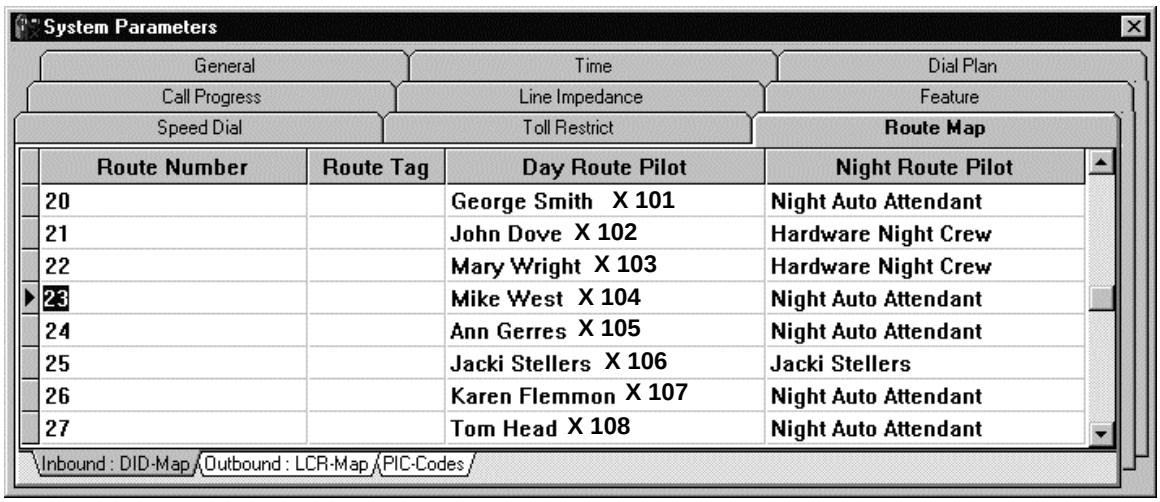


Table 13 - 13 System Parameters: Route Map: Inbound: DID-Map

Route Number	
The Caller ID number, DID number, DNIS number, or ANI number sent by the incoming service provider with the incoming call.	numeric, 1-20 digits
Route Tag	
Identifying name or information for this route number. This is used to identification or general information purposes only, it does not affect the call routing in any way.	alphanumeric, 0- 10 digits
Day Route Pilot	
System resource that the call will be routed to during the day mode.	None * User ID User Group ID AutoAttendant ID Voice ID
Night Route Pilot	
System resource that the call will be routed to during the night mode.	None * User ID User Group ID AutoAttendant ID Voice ID

* = default settings

14 Least Cost Routing

14.1 Overview

Least Cost Routing is a Plexus system feature which automatically routes external calls to the lowest cost route. The lowest cost route may be through accessing a specific service provider through any trunk or by accessing a trunk that has a pre-designated service provider. Least Cost Routing works by looking at information entered in the configuration and routing the calls based on this information. This can allow customers to have more than one service provider and to route calls to the lower cost provider based on the telephone number dialed and the time of day.

Note
It is the customers responsibility to determine and update the costs and rates for any service provider as the rates may change.

14.2 Description

The Least Cost Routing (LCR) feature is enabled on a per-user basis. When a user with LCR enabled dials the trunk access digit to make an external call, the Plexus system does not automatically assign a trunk to the call. The Plexus system will wait until the user has completed dialing the telephone number before determining the proper routing.

Once the user has completed dialing, the Plexus system compares the telephone number dialed to the LCR table. If the telephone number matches the numbers entered in the tables, the Plexus system routes the call according to the parameters entered in the tables. If the telephone number does not match the numbers entered in the tables, the call will be routed according to the user's default trunk assignment.

Note
The Plexus system buffers the digits dialed by the user until the user has completed dialing. The system waits for a set time after the last digit has been dialed to verify that no further digits will be dialed before routing the call. To reduce this wait time (default = 5 seconds), a user can press # when finished dialing. The Plexus system will then immediately route the call based on the defined settings.

14.3 Configuration

14.3.1 Overview

In order for LCR to work properly, several parameters must be configured. The parameters, working together, will route the outgoing call to the proper trunks and with the required PIC Codes (if used).

- User, Features, Options, Least Cost Routing Enabled,
- System Parameters: Time: Band
- System Parameters, Route Map, Outbound: LCR-Map,
- LCR-Map Definitions,
- PIC Codes

14.3.2 User, Features, Options, Least Cost Routing Enabled

Least Cost Routing is enabled on a per user basis. left-click in the box to enable Least Cost Routing for the user. Afterwards, right-click in the box to save time and Replicate the feature to All users or Selected users.

14.3.2.1 Interactions

A user must have User: Assignments: Off-Hook Preference defined as **None** (default) for LCR to work.

If a user does not have an assignment in User: Assignments: Trunk, external calls that do not match the numbers entered in the LCR tables will fail.

14.3.3 System Parameters: Time: Band

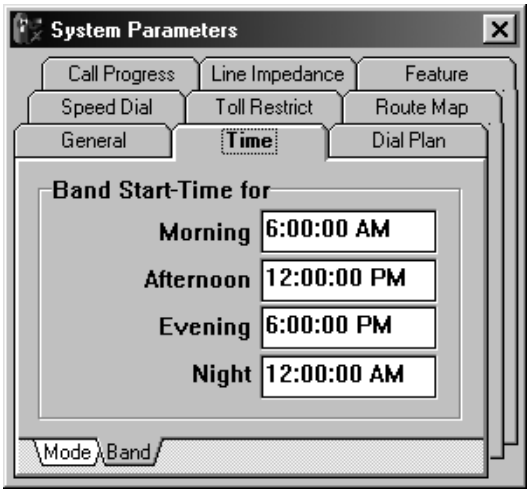


Table 14 - 1 System Parameters: Time: Band

Morning		
Time that Least Cost Routing starts following the Morning field in the LCR-Map Definition table to route calls.	Time; Hour, Minute, Seconds, AM/PM	6:00:00 AM*
Afternoon		
Time that Least Cost Routing starts following the Afternoon field in the LCR-Map Definition table to route calls.	Time; Hour, Minute, Seconds, AM/PM	12:00:00 PM*
Evening		
Time that Least Cost Routing starts following the Evening field in the LCR-Map Definition table to route calls.	Time; Hour, Minute, Seconds, AM/PM	6:00:00 PM*
Night		
Time that Least Cost Routing starts following the Night field in the LCR-Map Definition table to route calls.	Time; Hour, Minute, Seconds, AM/PM	12:00:00 AM*

* = default settings

14.3.4 System Parameters, Route Map, Outbound: LCR-Map

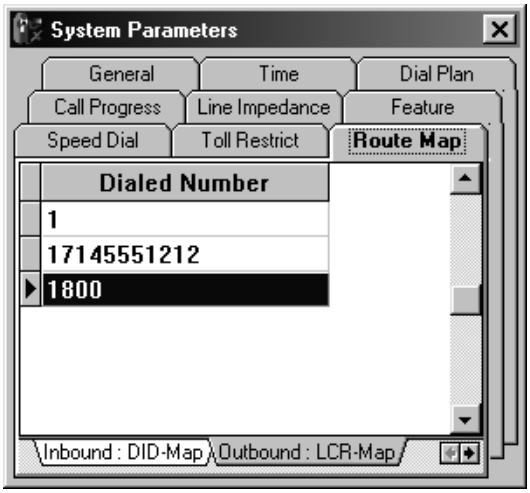


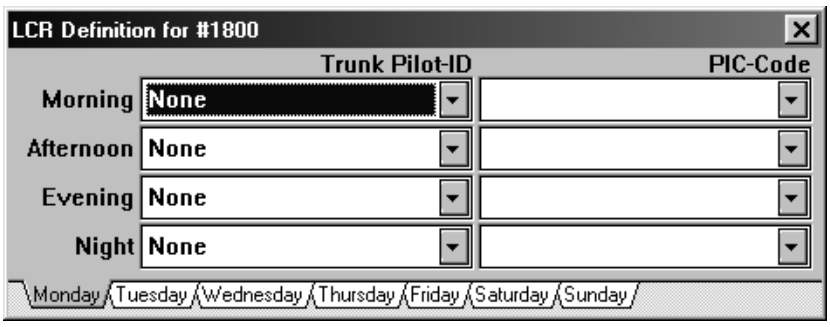
Table 14 - 2 System Parameters, Route Map, Outbound: LCR-Map

Dialed Number	
Number or prefix of the number dialed by a user with Least Cost Routing enabled.	Numeric, None*

* = default settings

14.3.5 LCR-Map Definitions

Double clicking in a completed Dialed Number field brings up the LCR-Map Definition window.



Tip
Right-clicking in any field gives you three time saving options: Replicate all times for same day - places this entry into all time fields on the chosen day, Replicate all days for same time - places this entry into the same time field for all seven days, and Replicate all numbers for day and time - places this entry into the same day and time field for all numbers entered in the Outbound: LCR-Map.

Table 14 - 3 LCR-Map Definitions

Trunk Pilot-ID	
Trunk or trunk group that will be accessed for this time band when the user dials the defined number.	None* Trunk ID Trunk Group ID
PIC-Code	
PIC Code that will be dialed with the number dialed by the user when the trunk is accessed.	Numeric, None*

* = default settings

14.3.6 PIC Codes

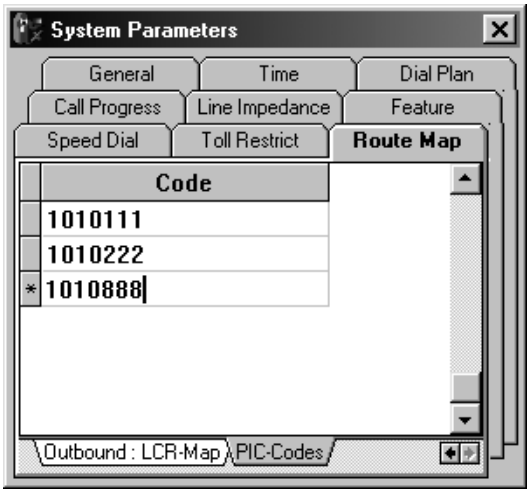
Primary Inter-Lata Carrier Codes (PIC Codes) are digits that are used by the system in conjunction with the user's dialed number to modify the dialed number when required. A PIC Code can be up to 20 digits in length.

14.3.6.1 PIC Codes - Basic Features

PIC Codes are digits that are sent to the service provider before sending the user's dialed digits. They are used to access a specific long-distance service provider without having that service provider as the main carrier on that line.

For example, if a customer subscribes to Company A for long-distance service but Company B offers lower rates during certain times, the customer could program the Plexus system to dial Company B's PIC Code before each long-distance call during that time. The call would be routed over the same trunk line but carried by and billed by Company B.

Note
PIC Codes are used to select a desired service provider. This is usually used for regional, national, or international long-distance calls, it does not apply to local calling.



Note
If any entries are modified or deleted, then any existing references in the LCR-Map Definitions are automatically cleared and must be redefined.

14.3.6.2 PIC Codes - Advanced Features

PIC Codes, as used in the Plexus systems, also offer flexibility in modifying dial strings for calling purposes. PIC Codes can add digits to the front of dialed numbers, strip numbers from the front of dialed numbers, or add numbers to the end of dialed numbers.

A. Add digits to the front of dialed number

Additional digits (0-9, *, #, or comma [2 second delay]) may be added to the front of dialed numbers. This is done by placing those digits in the Code field of PIC-Codes. This can be used for call routing or authorization through a service provider or for any purpose that may call for additional digits to be added.

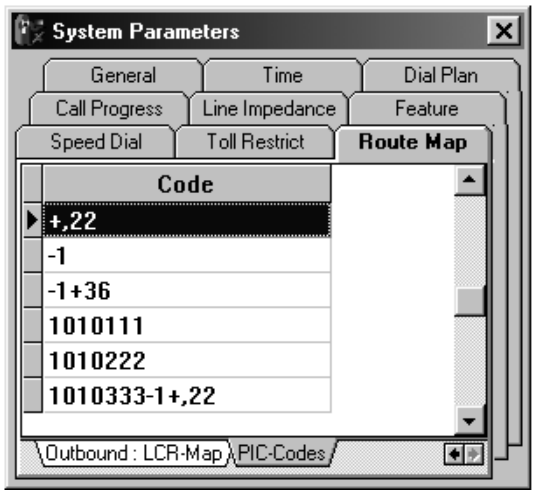
B. Strip digits from the front of dialed number

Under some conditions, it may be necessary to strip the first digit(s) from the dialed number before sending the digits to the service provider. This may be used with some digital trunks that do not require a 1 or 0 be dialed in long distance numbers or for other reasons. Entering a "-*n*" in the Code field of PIC-Codes will remove _ number of digits from the front of the dialed number. The system can remove between up to nine digits from the front of the dialed number.

If digits need to be added to the front of the dialed number, the "-*n*" must be placed after these digits (e.g., 1010888-1)

C. Add digits to the end of dialed number

Digits may be added to the end of the dialed number. This can be used for account numbers with service providers or for other reasons. Entering a "+*ddd*" at the end of the Code field of PIC-Codes will add the entered digits to the end or the user's dialed number. There can be up to 19 digits added to the end of a dialed number (the total length of the PIC Code field can not exceed 20 digits).



14.3.6.3 PIC Codes - Code - Overview

Table 14 - 4 PIC Codes - Code - Overview

Code	
Available PIC Codes that will be dialed with the number dialed by the user when the trunk is accessed with Least Cost Routing.	Up to 20 digits Numeric, +, -, *(star), #, comma, None

PIC Codes are entered in the following format:

PIC Code, *-n, +ddd* (e.g., 1010222,-1+22#)

Note
If the user dialed 1-800-555-1212 and the system used the example PIC Code (1010222,-1+22#). The system would end up dialing "1010222 <2 second pause> 800 555 1212 22#".

In most cases, the only digits needed are the actual PIC Code. In this case, the PIC Code would be entered in the following format:

PIC Code with no following digits (e.g., 1010222)

The following table defines the digits available for entry in PIC Codes:

Table 14 - 5 PIC Codes: Overview

Digit	Function	Acceptable Values
PIC Code	the number added to the front of the user's dialed number	0-9, *, #, comma
-n	subtracts *n* number of digits from the front of the user's dialed number	1-9
+ddd	adds the digits *ddd* (any number of digits) to the end of a user's dialed number	0-9, *, #, comma
#	dials a # in the dial string	# can be useful for terminating the dialed number string when accessing digital trunks or when providers have a dial timeout before routing a call.
*	dials a * in the dial string	
,(comma)	places a 2 second delay in the dial string	

14.4 Administration

While Least Cost Routing offers end-users a flexible way to reduce long-distance toll charges, maintaining too many service provides can actually increase maintenance costs and make the LCR tables overly complex. It is recommended that you limit the number of PIC Codes and entries in the Least Cost Routing tables.

15 Automated Attendant Interface

Note
This chapter is for use with the Auto Attendant Interface (AAI) card only. If you are configuring an Auto Attendant on an Integrated Voice Processor (IVP) card, do ***not*** use this chapter. Refer to chapter 9, "Integrated Voice Processor".

15.1 Introduction

This module explains how to install the optional Automated Attendant Interface (AAI) peripheral card and fully configure the Plexus auto attendant system.

15.1.1 Insert the AAI Peripheral Card

The AAI card is inserted into the system cabinet like any other peripheral card. Refer to chapter 4, "Installation" for complete information on peripheral card installation.

15.1.2 Adding an AAI Card to an Established Plexus System

When adding an AAI card to an established Plexus system, modifications will need to be made to the Plexus configuration file to account for the addition of the new peripheral card.

To modify the existing configuration:

1. Launch Plexus Administrator.
2. From the **File** menu, select **Open**.
3. Open the appropriate configuration file. Plexus configuration files have an extension of **.zdb**.
4. Add the AAI card to the existing configuration. Complete instructions for adding and deleting cards from the Plexus are available in chapter 6, "Software Configuration".

5 Configure the AAI card according to the instructions in the following pages.

6 After completing the configuration changes, the modified configuration file must be uploaded to the Plexus. Complete instructions for uploading configuration files are available in chapter 6, "Software Configuration."

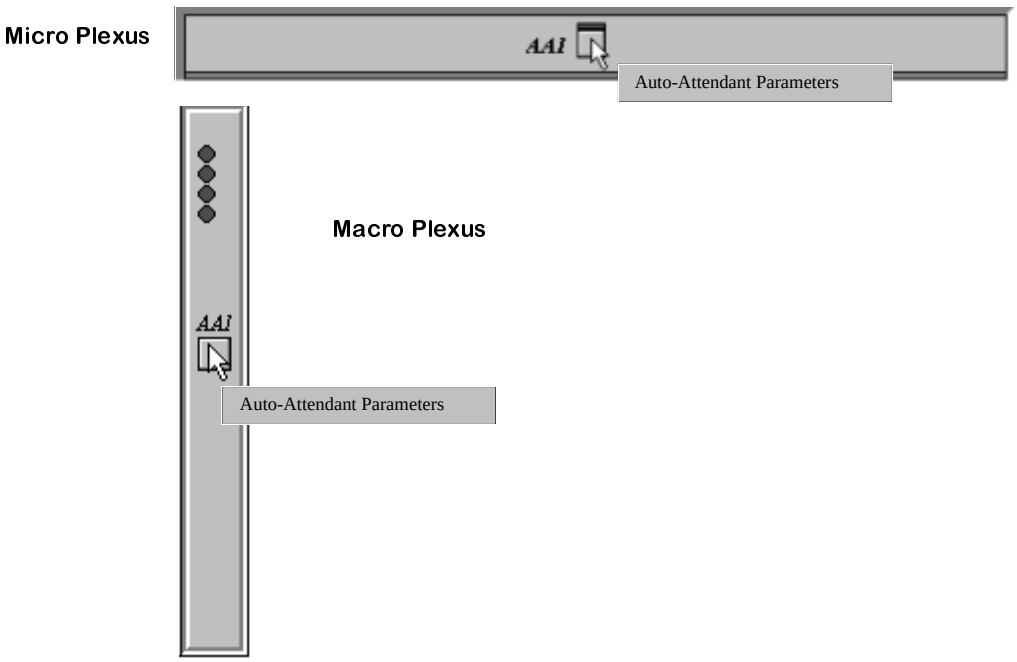
15.1.3 Incorporate an AAI Card with the Original Configuration

When incorporating an AAI card at the time of original install and configuration, follow the general instructions found in chapter 6, "Software Configuration." Specific information on configuring the AAI card is available in the remaining pages of the AAI manual.

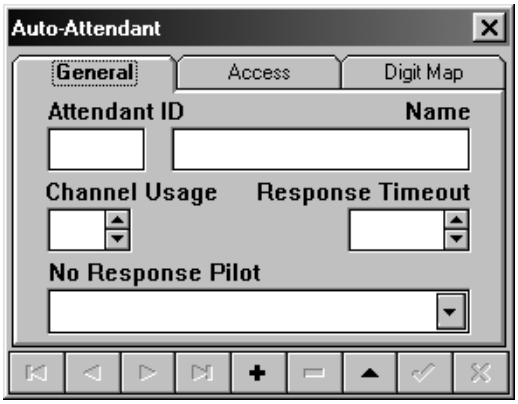
Refer to chapter 6, "Software Configuration" for instructions on card layout. An image of the AAI card must appear in the proper slot before the following configuration instructions can be carried out.

1 Click on the **System** tab within Plexus Administrator to display an image of the system cabinet and the available peripheral cards.

2 Click on the button on the image of the Auto Attendant Interface (AAI) card, as shown below.



3 The AAI dialog box will open with a number of configuration options.



4 Create and configure Auto Attendant IDs following the guidelines in the following sections.

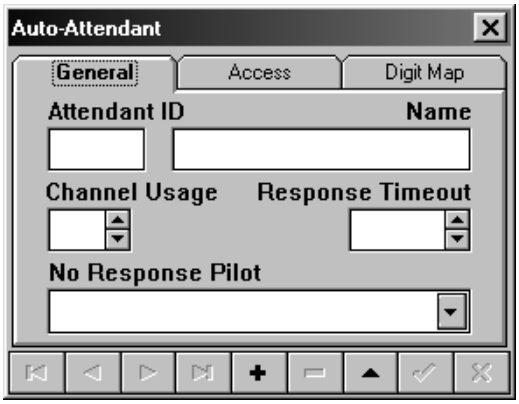
15.1.4 Navigating in Auto-Attendant Properties

The buttons and their functions are as follows:

⏮	First record	Scrolls to the first Auto-Attendant ID in the sequence.
◀	Prior record	Scrolls to the prior Auto-Attendant ID.
▶	Next record	Scrolls to the next Auto-Attendant ID.
⏭	Last record	Scrolls to the last Auto-Attendant ID in the sequence.
✚	Insert record	Inserts a new Auto-Attendant ID.
▬	Delete record	Deletes an existing Auto-Attendant ID.
▲	Edit record	Enables the parameters of a Auto-Attendant ID to be edited.
✓	Post Edit	Posts changes made to the Auto-Attendant ID parameters.
✕	Cancel Edit	Cancels changes made to the Auto-Attendant ID parameters.

15.1.5 Create Auto-Attendant IDs

1 After clicking on the button on the image of the AAI card, the following dialog window will be displayed.



2 To create an Auto-Attendant ID, click on the + button.

3 Enter a Auto-Attendant ID Sequence. The Auto-Attendant ID sequence is the start number of the range of logical IDs assigned to voice mail system resources.

4 The default settings call for a range from 1 to 4 with a width of 3. Therefore, the Auto-Attendant ID sequence may be any number between 100 and 499.

The Auto-Attendant ID sequence may begin with any number based on the **Dial-Number Range** system parameter. Refer to chapter 6, "Software Configuration" for more information on dial plans and dial-number ranges.

5 After entering a Auto-Attendant ID Sequence, additional Auto-Attendant IDs may be created by clicking on the + button.

6 Existing Auto-Attendant IDs may be edited from this dialog box by selecting the Edit button.

7 After editing an Auto-Attendant ID, the Post Edit button must be selected to record the changes. See “Navigating in Auto-Attendant Properties” - section 15.1.4 for more information about the Edit and Post Edit buttons.

8 Use the 9 buttons at the bottom of each Auto-Attendant tab to:

• scroll between existing Auto-Attendant IDs,

- create additional Auto-Attendant IDs according to the established sequence,
- edit and post changes to existing Auto-Attendant IDs, and
- delete existing Auto-Attendant IDs.

15.2 Configure Auto-Attendant IDs

The following sections provide information on configuring Auto-Attendant IDs:

15.2.1 General Properties

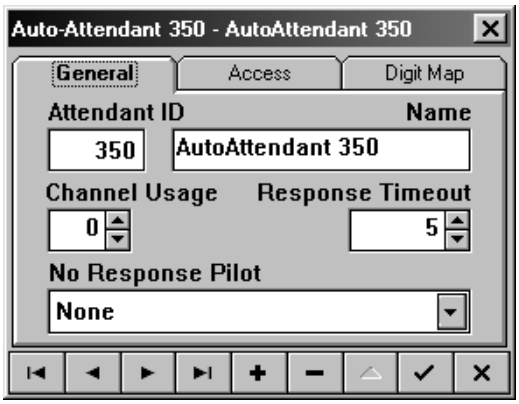


Table 15 - 1 General Properties

Attendant ID	
Identifier for this logical Auto-Attendant (e.g., 350).	Automatically Identified as ID that was entered in Create Auto Attendant ID
Name	
Name assigned to this logical Auto-Attendant (e.g., AutoAttendant 350). This name may be modified to something more meaningful, like “Main Greeting.”	Alphanumeric

Table 15 - 1 General Properties

Channel Usage	
Set Channel Usage to the number of channels that this auto-attendant will logically associate from the set of available physical channels. Each Auto-Attendant card provides four physical channels. Each physical channel may only be assigned to one Auto Attendant ID. For example, three physical channels may be assigned to the Day Greeting, and one physical channel may be assigned to the Night Greeting. Multiple Auto-Attendant cards may be utilized if more channels are required for a given logical Auto-Attendant. **Caution:** If the number of channels specified exceeds the number that are actually physically available, the system will be unable to complete the logical associations with the result that some logical Auto-Attendants will not be properly configured.	0-32, 0* **Caution:** If more than 4 channels are selected, there must be multiple Auto Attendant Interface cards on the system (4 channels per card).
Response Timeout	
Time that the system waits for a response following the end of the message. A response can be entered at any time *during* or *after* the message is played.	0-255 seconds, 5 seconds*
No Response Pilot	
This specifies where a call should be routed for further handling should no response be entered.	None* User User Group Trunk Trunk Group

* = default settings

Note
When using an AAI greeting for Intercept Service, make sure that the No Response Pilot is not defined. Otherwise, the call will be transferred to the pilot and out of the queue upon expiration of the Response Timeout.

For more information about Intercept Service, see chapter 8, "Feature Reference."

15.2.2 Access Properties

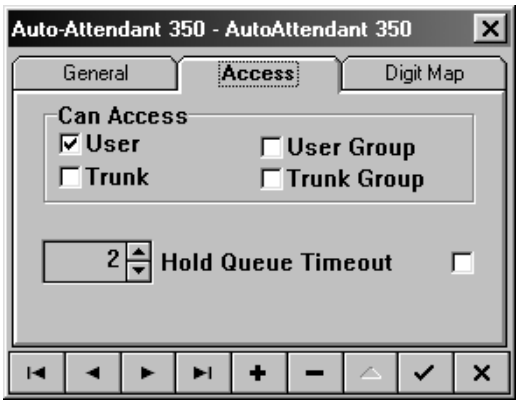


Table 15 - 2 Access

Can Access	
Configures what type of entities can be accessed via the Auto-Attendant.	User * User Group Trunk Trunk Group
Hold Queue Timeout	
If queuing is enabled and all the channels associated with this logical auto-attendant are busy, the system will queue the call until a channel is available or the Hold Queue Timeout expires.	Enabled* 0-255 minutes

* =default settings

Digit Map Properties

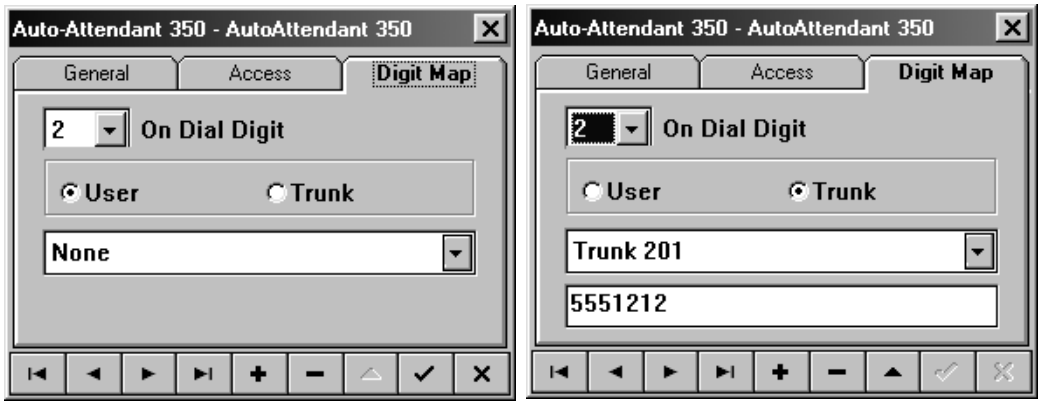


Table 15 - 3 Digit Map

On Dial Digit	
Specifies the dial digit that will be associated with the Pilot selected below.	0-9, #, *(star)
User / Trunk	
Specifies whether the call will be routed to a user / user group or a trunk / trunk group. Each digit (0 to 9, # and *) can have a unique pilot assigned to it.	None * User ID User Group ID Trunk ID Trunk Group ID
Externally Dialed Number	
If Trunk is selected in the field above, another field appears for the external number that the call will be transferred to.	up to 20 digits - 0-9, #, *(star), (,) comma [2 second delay]

* =default settings

15.2.3 Record a Message Greeting

Before recording a message greeting, please make note of the following points:

- The maximum length that a message recording can be is 20 seconds.

- Your message will be recorded simultaneously on all the physical channels that were associated with the specified logical Auto-Attendant.
- If the number of channels associated with an Auto-Attendant is subsequently modified, you may need to re-record all your messages

To record a message greeting from either a Plexus key telephone or an ordinary phone:

1. Activate System Administrator Privileges.
2. Enter **<F96>** + Auto Attendant ID.
3. Record a greeting. See below for sample greetings.
4. To finish recording the greeting, hang-up gently or press the **SPKR** button on key telephones.

3OD\ D 5HFRUGHG 0HVVDJH *UHHWLQJ

To play a recorded message greeting, simply dial the Auto Attendant ID from either a Plexus key telephone or an ordinary phone. When the Auto-Attendant is activated in this manner, all the routing is fully functional. This is a good way to quickly verify that your settings are working as desired.

6DPSOH 0HVVDJH *UHHWLQJV

A. Day Greeting

"Thank you for calling [***your company name***]. If you know your party's extension, you may dial it at any time. For Technical Support, press 5. For the Sales Department, press 6. For Accounting, press 7. For Purchasing, press 8. For personal assistance, press 0, or hold the line."

B. Night Greeting

"Thank you for calling [***your company name***]. Our office hours are from 8:00 a.m. to 6:00 p.m., Monday through Friday. Please contact us during our normal operating hours. Thank you."

C. Intercept Greeting

"Please stay on the line. The next available operator will be with you in a moment."

&7, ,QWHUIDFH

2YHUYLHZ

BBS Telecom is developing a Computer-Telephony Interface (CTI) solution aimed at small centers. The Plexus Telephone System uses the new CTI-version of the DXP card to provide third party call control using CSTA-standard commands. Dialogic's CT Connect provides high level TAPI, TSAPI, and CT Connect APIs and is a required part of the solution.

Please contact your BBS Telecom representative for further details on the Plexus CTI solution.

5HPRWH 6\VWHP 0DQDJHPHQW

$ERXW 5HPRWH 6\VWHP 0DQDJHPHQW

The Remote System Management feature of Plexus Administrator enables a Plexus system to be configured and otherwise managed via modem connection. In a Remote System Management environment, the PC running Plexus Administrator is connected to the Plexus through a modem as opposed to a direct serial connection.

Remote System Management requires a modem at both locations. One modem, the System Modem, needs to be connected to the Plexus system. The other modem, the Manager Modem, is connected to the PC running Plexus Administrator. From the PC, a call is placed through the Manager Modem to the trunk (i.e., CO line) to which the System Modem is connected. Once connected, a link is opened just as it would be if the PC were connected directly to the Plexus system with a serial cable. All functionality available with a direct serial connection is available with a modem connection

Note
It is recommended that a modem be hooked up to an analog trunk. This will enable Remote System Management even in the event of a systemwide failure. Connections to extensions will not allow this. XOS software upgrading is also unavailable with a connection through an extension.

5HTXLUHPHQWV

Remote Configuration Kits are available through BBS Telecom. The Remote Configuration Kit includes:

1 System Modem- Dip switches are preset by BBS Telecom

2 110V modem power supply

3 null modem cable

To successfully complete Remote System Management functions, certain requirements must be met. The requirements are:

- An external modem (System Modem) connected to the Plexus system according to the instructions below.
- A null modem cable.

Note
Null modem cables are provided by BBS Telecom with Remote Configuration Kits.

- A second modem (Manager Modem - dealer provided). This modem must be connected to the PC running Plexus Administrator. The modem must be 19.2 Kbps or faster.
- A telephone line connected to the Manager Modem.

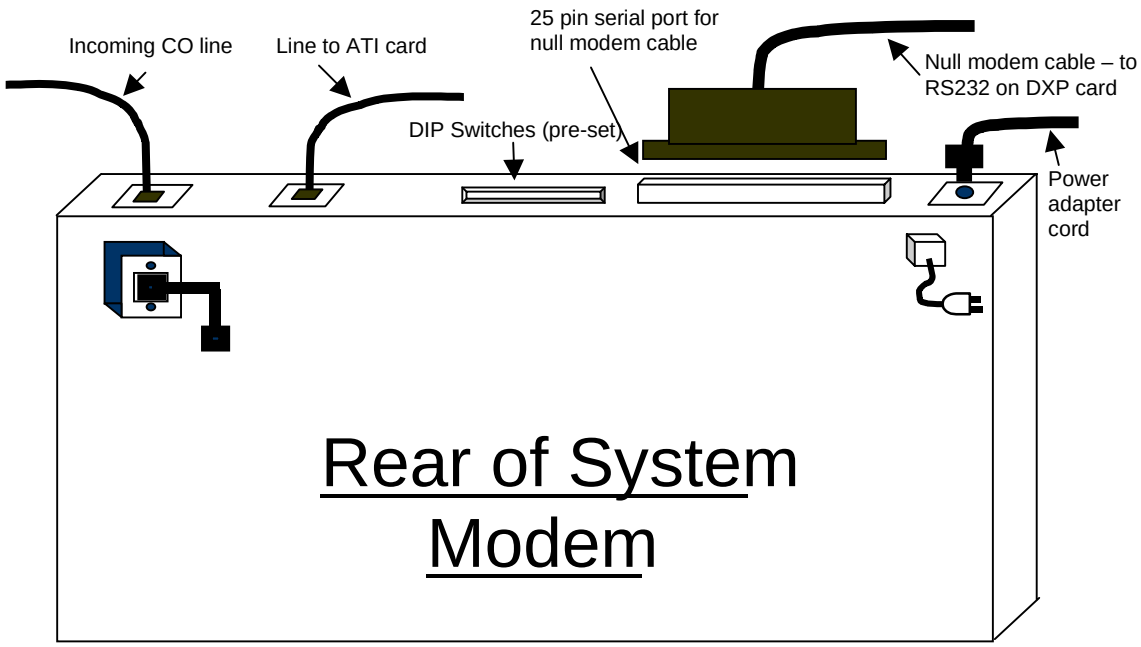
3OH[XV 6\VWHP VHWXS

Plexus systems that will be remotely managed must be set up as follows:

1 Connect the incoming CO line to the incoming line port of the System Modem.

2 Connect a line cord from the outgoing line port on the System Modem to a port on the ATI card on the Plexus. This step is not necessary if there is going to be a dedicated CO line connected to the System Modem and the CO line will not be available for Plexus system use.

Figure 17.1



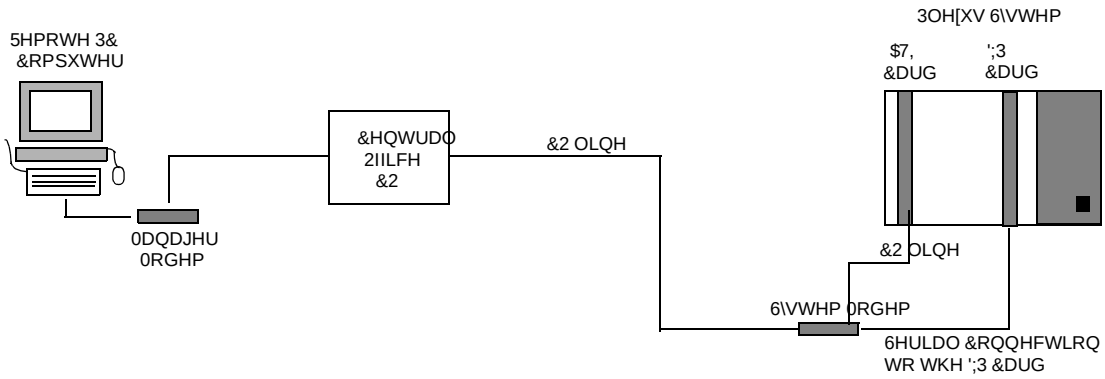
3 Connect the serial port on the DXP card (labeled “RS-232”) to the serial port on the System Modem using the null modem cable provided by BBS Telecom.

4 Plug the System Modem in to a power source.

5 Turn the modem OFF when not in use.

6 The modem setup should resemble the diagrams if figure 17.1 and figure 17.2.

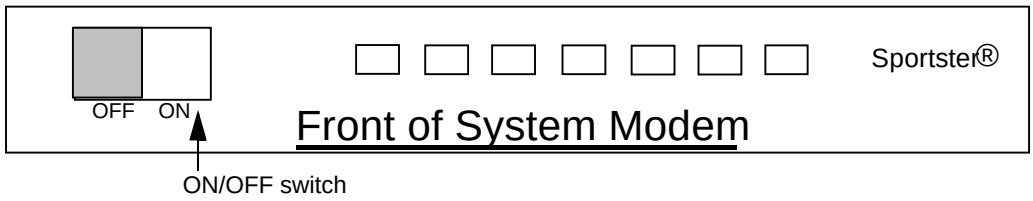
Figure 17.2



7 When the modem is not in use, it should be turned off.

8 When the modem will be needed, have a person on site turn the modem ON using the switch shown in figure 17.3.

Figure 17.3



3OH[XV $GPLQLVWUDWRU VHWXS

Refer to Chapter 6, "Software Configuration" for instructions on launching Plexus Administrator.

1 Launch Plexus Administrator.

2 From the Link menu, select Setup and Port .

3 Select the COM port to which the Manager Modem is connected.

Note
The Manager Modem may be any make or model modem, and may be either internal or external. The modem speed must be 19.2 Kbps or faster.

&RQQHFWLQJ WR WKH 3OH[XV V\VWHP

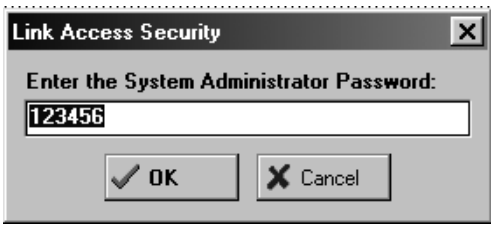
To connect to the Plexus system, proceed as follows:

1 Have someone at the Plexus system site turn the System Modem ON.

2 From the Link menu, select Setup, Connection, and Modem.

3 From the Link menu, select Open.

4 You will be prompted to enter a password. This will be the System Administrator Password for the system you are dialing into.



5 Enter the password and click OK.

6 The Modem Setup Parameter window will be displayed.

7 On the Modem Setup Parameters dialog box, enter the appropriate Modem Initialization Sequence and Telephone Number.

Note
The modem initialization sequence will not need to be modified for most modems 33.6k speed or slower. If your modem is a 56k modem, refer to your modems documentation for the modem initialization string that will slow your modem to a 33.6k speed or slower. This should be placed before the AT&F portion of the modem initialization sequence.

8 Select the Dial Mode .

9 Click on Connect

10 Plexus Administrator will dial the System Modem and connect.

11 Once connected the bottom left portion of the Modem Setup Parameters dialog will indicate Connected

12 Once connected, click on OK.

13 An active link is now available.

14 The link indicator at the bottom of the screen should indicate Link: Opened and the LED image should appear green.

15 Verify the connection by clicking on the magnifying glass icon. This will query the current XOS version. If a reply is received and the XOS version appears in a pop-up window, the connection is successful.

Note
If a configuration is created or modified and saved while using Remote System Management, the dialed number will be saved with the configuration file.

8SORDGLQJ D FRQILJXUDWLRQ ILOH

With an established connection and an open link, proceed as follows:

1 Open the applicable configuration file (.zdb).

2 From the Update menu, select Full.

3 A status indicator at the bottom of the screen will display the progress of the update.

Tip
In some cases, it may be preferable to select Update Clear prior to selecting Update Full. This action clears the existing configuration from the *Plexus* system.

8SGDWLQJ WKH V\VWHP FORFN

With an established connection and an open link, proceed as follows:

1 Open any configuration file (.zdb).

Note
A configuration file must be open for the Update menu to be available. The open configuration file will not be uploaded or used in any way to update the clock.

2 From the Update menu, select Clock .

3 The Plexus system clock will be updated with the clock settings from the PC.

8SGDWLQJ 3OH[XV .H\ 7HOHSKRQH SURJUDPPLQJ

17.8.1 For Version 1.01 Key Telephones

These steps are used with Key Telephones version 1.01 only.

With an established connection and an open link, proceed as follows:

1 Open the applicable configuration file (.zdb).

2 From the Update menu, select Key-Phone

3 A status indicator at the bottom of the screen will display the progress of the update.

Note
Updating Key Telephones through the Update - Key Phone option will cause each version 1.01 Key Telephone to reset, disconnecting active calls.

17.8.2 For Version 2.0 Key Telephones or later

These steps are used with Key Telephones version 2.0 and later only.

With an established connection and an open link, proceed as follows:

1 Open the applicable configuration file (.zdb).

2 Open the window User: Features: Settings: Define Key for the user's Key Telephone being programmed.

3 Complete the Key Telephone programming for the user.

4 Press Update button. This will update only this user's Key Telephone.

Note
The Update now button will not be an available option unless the window was opened while having an active link with the system.

or to program all Key Telephones

After completing the key telephone programming (for all of the users on the system), and with the configuration file open:

1 From the Link menu, select Open .

2 The link indicator at the bottom of the screen should indicate Link: Opened and the LED image should appear green

3 From the Update menu, select Key-Phone .

Note
Updating Key Telephones through the Update - Key Phone option will cause each version 1.01 Key Telephone to reset, disconnecting active calls.

&ORVLQJ WKH OLQN DQG GLVFRQQHFWLQJ

With an established connection and an open link, proceed as follows:

1 From the Link menu, select Close

Hotel & Motel Package

18

18.1 Introduction

The Plexus Hybrid-Key Digital Telephone System offers a set of feature codes and Operator Console filters designed specifically for the Hotel and Motel industry. No additional hardware or software is required to utilize the feature set. However, the station where the feature codes will be issued must have System Administrator Privileges activated. A Plexus DSS-60/DDSS-60 Operator Console is necessary to utilize the filters which provide room status information at a glance.

Another Feature designed especially for the Hotel and Motel industry is the availability of "simple" voice mailboxes if the system has an Integrated Voice Processor (IVP) interface card installed. These are mailboxes designed to interact with the feature codes and to block the room occupant's ability to modify the voice mailbox settings.

18.1.1 About This Module

This optional module for the Plexus System Manual addresses each of the Hotel and Motel features and all of the Operator Console filters. In preparation for using the various Hotel and Motel features, instructions for activating System Administrator Privileges and programming the DSS-60/DDSS-60 Operator Console are presented below.

18.2 System Administrator Privileges

It is necessary to have System Administrator Privileges activated prior to issuing any Hotel and Motel feature code. To simplify the process of issuing Hotel and Motel feature codes, it is recommended that certain secure stations on the PBX be designated for this function (e.g., Front Desk phone). If this strategy is adopted, System Administrator Privileges can be activated once at the designated station(s) and kept active.

It is possible to issue a Hotel and Motel feature code from any station on the system. However, it will be necessary to first log into the station as a user with System Administrator Privileges and then activate that the privileges. When issuing feature codes from a non-secure station (e.g., in a guestroom), it is advisable that the original user ID be logged back onto the station and that System Administrator Privileges be deactivated immediately following the issuance of the feature code.

A. To activate System Administrator Privileges:

When the system is set, users with System Administrator Privileges will have the privileges activated by default.

or

If the System Administrator Privileges have been deactivated,

dial **F91** + System Administrator Password

Note
The default password is **123456**.

B. To deactivate System Administrator Privileges:

dial **F90**

18.3 DSS-60/DDSS-60 Operator Console

Many of the Hotel and Motel feature codes activate informative indicators on a properly installed and programmed DSS-60 Analog Operator Console or DDSS-60 Digital Operator Console. These indicators provide information about each guestroom, such as whether the room is occupied, in need of service, has messages, or is in Do-Not-Disturb (DND) mode. To view such information, each room is assigned a key on the Operator Console(s). The LED on the key will illuminate or flash under various circumstances to indicate a particular status (e.g., Occupied). Different room status information is available depending on the active filter.

18.3.1 Filters

Several filters are available for viewing different room status information. Each filter has a corresponding Filter Code. The Filter Codes are assigned to Operator Console keys (see **Programming the DSS-60/DDSS-60 Operator Console(s)**). Pressing these keys activates the different filters thereby displaying different information about the guestrooms.

For example, if the "Occupied" filter is active, only the keys for occupied rooms will be lit.

The following fourteen (14) Filter Codes are available:

Table 18 - 1 DSS-60/DDSS-60 Operator Console: Filter Codes

Code	Name	Lights indicator if the guestroom...
1	Message Waiting	has messages waiting.
2	Do-Not-Disturb	is in DND mode.
3	No Outside Line Access	cannot access outside lines.
4	No Room Access	cannot access other room phones.
5	Occupied	is occupied.
6	Not Occupied	is not occupied.
7	Serviced	is serviced.
8	Not Occupied, Serviced	is not occupied and is serviced

Table 18 - 1 DSS-60/DDSS-60 Operator Console: Filter Codes

9	Not Occupied, Not Serviced	is not occupied and is not serviced.
10	Occupied, Serviced	is occupied and is serviced.
11	Occupied, Not Serviced	is occupied and is not serviced.
12	Not Serviced	is not serviced.
13	Wake-Up Failed	did not answer wake-up call
14	Left Off Hook	has left phone off hook

18.3.2 Installation Considerations

18.3.2.1 DSS-60 Analog Operator Console

The DSS-60 Analog Operator Console is installed in the same manner as any other extension device (e.g., *Plexus* Analog Key Telephone, ordinary phone, etc.). Each DSS-60 utilizes an extension port on an Analog Extension Interface (AEI) card. Therefore, separate station wiring must be available for each Analog Operator Console.

To facilitate programming, each DSS-60 must have a host Plexus Analog Key Telephone (i.e., PVT-30D). Each host is capable of programming up to seven (7) DSS-60 Operator Consoles. All of the DSS-60 Analog Operator Consoles must be connected to the same AEI card as the host analog key telephone.

Note
The DSS-60 analog console only operates with the Plexus Analog Key Telephones of the same software version number (e.g., 1.01 or 2.0).

18.3.2.2 DDSS-60 Digital Operator Console

The DDSS-60 Digital Operator Console is installed in the same manner as the PDT series digital key telephones. Each DDSS-60 utilizes an extension port on a Digital Extension Interface (DEI) card. Therefore, separate station wiring must be available for each Digital Operator Console.

To facilitate programming, each DDSS-60 must have a host Plexus Digital Key Telephone (i.e., PDT-30D). Each host is capable of programming up to seven (7) DDSS-60 Digital Operator Consoles. All of the DDSS-60 Digital Operator Consoles must be connected to the same DEI card as the host digital key telephone.

Note
The DDSS-60 digital console only operates with the Plexus Digital Key Telephone.

18.3.3 Configuration Considerations

Operator Consoles should be assigned sequential User IDs based on the User ID (i.e., extension number) of the host key telephone.

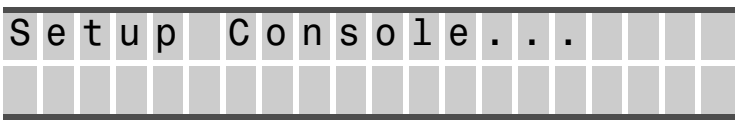
If the host key telephone is assigned a User ID of 150, the related DSS-60/DDSS-60 Operator Console(s) should have sequential User IDs beginning with 151.

18.3.4 Programming the DSS-60/DDSS-60 Operator Console(s)

Once installed and configured, the DSS-60/DDSS-60 must be programmed using the host key telephone. It is recommended that each guestroom have an assigned key on the DSS-60/DDSS-60 Operator Console(s). These keys allow hotel personnel to monitor phone activity and view room status indicators using the various filters.

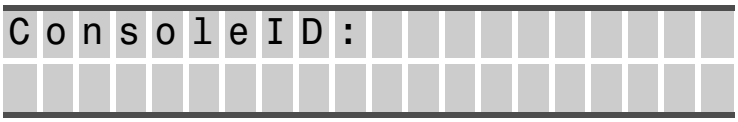
18.3.4.1 Assigning Room Keys

1. On the host key telephone, press and hold down **OVERRIDE** for 2 seconds.
2. Press **VOL** ▲ until the option **Setup Console** appears.



3. Press **SPKR** to select this mode.

4 At the **Console ID** prompt enter the User ID that has been assigned to the desired Operator Console.

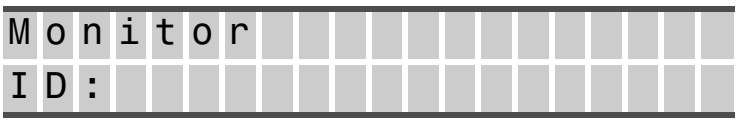


5 Press **SPKR**.

All the programmable keys on the corresponding Operator Console will illuminate.

6 On the Operator Console, press the Operator Console key to be programmed.

7 On the host key telephone, press **VOL ▲** until the desired parameter appears.



8 Operator Consoles can have each key used to monitoring extensions or trunks, provide access to features or functions, or as filter keys for special applications (e.g., Hotel and Motel).

9 For Hotel and Motel applications, use these keys to monitor the User IDs (i.e., room number) of the rooms.

10 Press the next key to program.

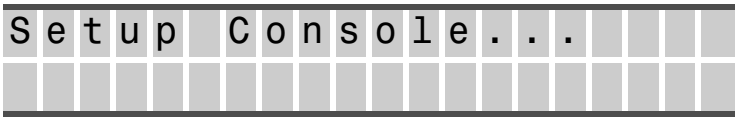
11 Repeat steps 6-10 until each room has a defined key.

Tip
Label each key on the DSS-60/DDSS-60 with the appropriate room number. Pre-cut overlay templates and software to customize the labels are available from dealers or Customer Service [Refer to the **Customer Service** section of the *Plexus* System Manual].

18.3.5 Assigning Filter Keys

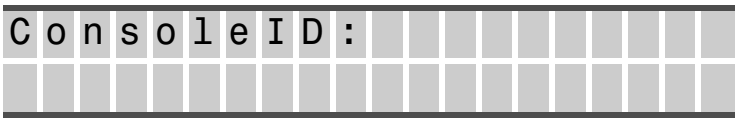
1 On the host key telephone, press and hold down **OVERRIDE** for 2 seconds.

2 Press **VOL** ▲ until the option **Setup Console** appears.



3 Press **SPKR** to select this mode.

4 At the **Console ID** prompt enter the User ID that has been assigned to the desired Operator Console.



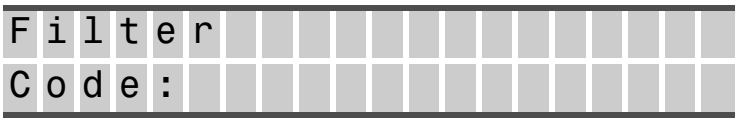
5 At the "**Console ID :**" prompt enter the User ID that has been assigned to the desired Operator Console.

6 Press **SPKR**.

All the programmable keys on the corresponding Operator Console will illuminate.

7 On the Operator Console, press the Operator Console key to be programmed.

8 On the host key telephone, press **VOL** ▲ until the desired parameter appears.



9 Enter the desired Filter Code.

10 Press the next key to program.

11 Repeat steps 6-10 until each desired filter key has been programmed.

In order to utilize filters, the appropriate filter keys need to be programmed on each DSS-60/DDSS-60 Operator Console in use.

Note
Reserve the bottom row(s) of keys on each DSS-60/DDSS-60 for filter keys depending on the number of filters that will be utilized.

18.4 Simple Voice Mailboxes.

Simple voice mailboxes are individual mailboxes that give the users (room occupants) no access to ordinary mailbox features but let the users listen to voice messages, save voice messages, or delete voice messages. This ensures that settings can not be modified by each user. A generic greeting can be recorded in the mailbox by the system administrator. For further details on setting up a simple voice mailbox, refer to chapter 9, "Integrated Voice Processor."

Simple voice mailboxes interact with the feature codes. When a guest checks out of a room, the feature for guest checkout is entered for that room. The system will delete all messages saved in the room's voice mailbox to ready the voice mailbox for the next guest.

18.5 Features

18.5.1 Check-In

To reset a guestroom phone and designate the guestroom as "Occupied," front desk personnel should issue the following code upon guest check-in. The filters activated by this code enable hotel personnel to determine room availability at a glance. Activated filters may also highlight the need for a room to be quickly serviced (e.g., in the case where a guest is checked into a room designated as 'Not Serviced').

18.5.1.1 Code

F75 + Room No.

The Check-In code will...

- Establish that the guestroom is occupied.
- Enable access to outside lines.
- Enable access to other room phones.
- Clear Do-Not-Disturb (DND) mode.
- Clear the Message Waiting indicator and clear the room's voice mailbox (if applicable) .
- Clear any failed alarm indicator.

18.5.1.2 Filter Interactions

Table 18 - 2 Features: Check-In: Filter Interactions

Code	Name	Action
0	Message Waiting	deactivate
1	Wake-Up Failed Alarm	deactivate
3	DND	deactivate
4	No Outside Line Access	deactivate
5	No Room Access	deactivate
6	Occupied	activate
7	Not Occupied	deactivate
11	Occupied, Serviced	activate
12	Occupied, Not Serviced	activate

18.5.1.3 Feature Interactions

The Check-In code will either activate the "Occupied, Serviced" or the "Occupied, Not Serviced" filter depending on the last Room Service Indicator code issued. In other words, if Housekeeping issued a **F78** deactivate code subsequent to a Check-Out code, the Check-In code would activate the "Occupied, Serviced" filter.

18.5.2 Check-Out

To disable a guestroom phone and designate the guestroom as "Not Occupied, Not Serviced," front desk personnel should issue the following code upon guest check-out. The filters activated by this code enable hotel personnel to determine room availability at a glance. Activated filters also highlight the need for the room to be serviced.

18.5.2.1 Code

F74 + Room No.

The Check-Out code will:

- Establish that the guestroom is unoccupied.
- Establish that the room needs to be serviced.
- Disable access to outside lines.
- Disable access to other room phones.
- Clear Do-Not-Disturb (DND) mode.
- Clear the Message Waiting indicator and clear the room's voice mailbox (if applicable) .
- Clear any failed alarm indicator.

18.5.2.2 Filter Interactions

Table 18 - 3 Features: Check-Out: Filter Interactions

Code	Name	Action
0	Message Waiting	deactivate
1	Wake Failed	deactivate
3	DND	deactivate
4	No Outside Line Access	activate
5	No Room Access	activate
6	Occupied	deactivate
7	Not Occupied	activate
8	Serviced	deactivate
11	Not Occupied, Not Serviced	activate

18.5.2.3 Feature Interactions

The Check-Out code always activate the "Not Occupied, Not Serviced" indicator. This will highlight the need for Housekeeping to service the room. Once the room is serviced, a **F78** code should be issued to indicate that the room has been serviced.

18.5.3 Message Waiting

When a message is taken for a hotel guest, hotel personnel may activate the message waiting lamp in the appropriate guestroom. There are feature codes available to activate and deactivate the message waiting lamp in each guestroom.

18.5.3.1 Activate Code

F51 + Room No.

18.5.3.2 Filter Interactions

Table 18 - 4 Features: Message Waiting: Filter Interactions

Code	Name	Action
0	Message Waiting	activate

18.5.3.3 Deactivate Code

F50 + Room No.

18.5.3.4 Filter Interactions

Table 18 - 5 Features: Message Waiting: Filter Interactions

Code	Name	Action
0	Message Waiting	deactivate

18.5.3.5 Feature Interactions

Check-In and Check-Out codes automatically deactivate the message waiting lamp in the appropriate guestroom and the related Operator Console indicator.

18.5.4 Do Not Disturb

Should a guest request that they not be disturbed, hotel personnel may activate Do-Not-Disturb (DND) mode on the guest's phone. There are feature codes available to activate and deactivate DND mode.

18.5.4.1 Activate Code

F73 + Room No.

18.5.4.2 Filter Interactions

Table 18 - 6 Features: Do-Not-Disturb: Filter Interactions

Code	Name	Action
3	DND	activate

18.5.4.3 Deactivate Code

F72 + Room No.

18.5.4.4 Filter Interactions

Table 18 - 7 Features: Do-Not-Disturb: Filter Interactions

Code	Name	Action
3	DND	deactivate

18.5.4.5 Feature Interactions

Check-In and Check-Out codes automatically deactivate DND mode in the appropriate guestroom and the related Operator Console indicator.

18.5.5 Room Occupancy Indicator

While Check-In and Check-Out codes automatically activate the appropriate occupancy indicators on the various filters, the following feature codes are available to manually activate or deactivate the occupancy indicators.

Note
Upon issuing a Check-In code, the guestroom will automatically appear as ‘Occupied’ when the applicable filters are utilized at the Operator Console. Upon issuing a Check-Out code, the guestroom will automatically appear as "Not Occupied" when the applicable filters are utilized at the Operator Console.

18.5.5.1 Activate Code

F77 + Room No.

18.5.5.2 Filter Interactions

Table 18 - 8 Features: Room Occupancy Indicator: Filter Interactions

Code	Name	Action
6	Occupied	activate
7	Not Occupied	deactivate
9	Not Occupied, Serviced	deactivate
10	Not Occupied, Not Serviced	deactivate
11	Occupied, Serviced	activate
12	Occupied, Not Serviced	activate

18.5.5.3 Deactivate Code

F76 + Room No.

18.5.5.4 Filter Interactions

Table 18 - 9 Features: Room Occupancy Indicator: Filter Interactions

Code	Name	Action
6	Occupied	deactivate
7	Not Occupied	activate
9	Not Occupied, Serviced	activate
10	Not Occupied, Not Serviced	activate
11	Occupied, Serviced	deactivate
12	Occupied, Not Serviced	deactivate

18.5.5.5 Feature Interactions

- Check-In and Check-Out codes automatically activate the appropriate occupancy indicator.
- The Room Occupancy Indicator codes will activate the appropriate "Serviced" or "Not Serviced" indicator depending on the current status activated from a Check-Out or Room Service Indicator code (i.e., **F78** or **F79**).

18.5.6 Room Service Indicator

While the Check-Out code automatically activates the "Not Serviced" indicator on the various filters, the following feature codes are available to manually activate or deactivate the service indicators. These feature codes may be used to alert hotel personnel when a room is in need of service. Housekeeping or Maintenance personnel may issue the deactivate code upon servicing a room.

Note
Upon issuing a Check-Out code, the guest room automatically appears as "Not Serviced" when the applicable filters are utilized at the Operator Console. Issuing a Check-In code does not automatically assign "Serviced" or "Not Serviced" to the guest room (as the room may be either "Serviced" or "Not Serviced" upon Check-In).

18.5.6.1 Activate Code

F79 + Room No.

18.5.6.2 Filter Interactions

Table 18 - 10 Features: Room Service Indicator: Filter Interactions

Code	Name	Action
8	Serviced	deactivate
9	Not Occupied, Serviced	deactivate
10	Not Occupied, Not Serviced	activate
11	Occupied, Serviced	deactivate
12	Occupied, Not Serviced	activate

18.5.6.3 Deactivate Code

F78 + Room No.

18.5.6.4 Filter Interactions

Table 18 - 11 Features: Room Service Indicator: Filter Interactions

Code	Name	Action
8	Serviced	activate
9	Not Occupied, Serviced	activate
10	Not Occupied, Not Serviced	deactivate
11	Occupied, Serviced	activate
12	Occupied, Not Serviced	deactivate

18.5.6.5 Feature Interactions

- Check-Out codes automatically activate the "Not Serviced" indicator.
- The Room Service Indicator codes will activate the appropriate "Occupied" or "Not Occupied" indicator depending on the current status activated from a Check-In, Check-Out, or Room Occupancy Indicator code (i.e., **F76** or **F77**).

18.5.7 Access to Outside Lines

While Check-In and Check-Out codes automatically enable/disable access to outside phone lines from the guest room, the following feature codes are available to manually enable or disable access.

Note
Upon issuing a Check-Out code, access to outside phone lines from the guest room is automatically disabled. Upon issuing a Check-In code, access to outside lines from the guest room is automatically enabled.

18.5.7.1 Activate Code

F83 + Room No.

18.5.7.2 Filter Interactions

Table 18 - 12 Features: Access to Outside Lines: Filter Interactions

Code	Name	Action
4	No Outside Line Access	deactivate

18.5.7.3 Deactivate Code

F82 + Room No.

18.5.7.4 Filter Interactions

Table 18 - 13 Features: Access to Outside Lines: Filter Interactions

Code	Name	Action
4	No Outside Line Access	activate

18.5.7.5 Feature Interactions

- Check-In codes automatically enable access to outside lines and deactivate the related Operator Console indicator.
- Check-Out codes automatically disable access to outside lines and activate the related Operator Console indicator.

18.5.8 Access to Other Rooms

While Check-In and Check-Out codes automatically enable/disable access to other room phones from the guest room, the following feature codes are available to manually enable or disable access.

Note
Remember: Upon issuing a Check-Out code, access to other room phones from the guest room is automatically disabled. Upon issuing a Check-In code, access to other room phones from the guest room is automatically enabled.

18.5.8.1 Activate Code

F85 + Room No.

18.5.8.2 Filter Interactions

Table 18 - 14 Features: Access to Other Rooms: Filter Interactions

Code	Name	Action
5	No Room Access	deactivate

18.5.8.3 Deactivate Code

F84 + Room No.

18.5.8.4 Filter Interactions

Table 18 - 15 Features: Access to Other Rooms: Filter Interactions

Code	Name	Action
5	No Room Access	activate

18.5.8.5 Feature Interactions

- Check-In codes automatically enable access to other room phones and deactivate the related Operator Console indicator.
- Check-Out codes automatically disable access to other room phones and activate the related Operator Console indicator.

18.5.9 Wake-Up Calls

Hotel personnel can program a guestroom phone to ring at a specified time as an alarm, notification, or wake-up call. Should the guestroom phone be unavailable (i.e., busy or in DND mode) or if the guest does not answer, the system can retry the room. The System Administrator can program the number of retries that the system will attempt and the duration (in minutes) between retries.

The System Administrator can also define a station (e.g., the Front Desk phone) to serve as the No-Response Cover Pilot [Refer to the **Features** (Alarm Call) tab under System Parameters]. The No-Response Cover Pilot receives a failed alarm call when all retries of a wake-up call fail to reach the guest. Failed alarm calls generate stutter tone followed by Music-On-Hold. When Front Desk personnel receive a failed alarm call they should examine the Operator Console to determine which room experienced a failed alarm. When an alarm fails for a given room the indicator lamp for that room will flash rapidly [No filter key is necessary to view failed alarm indicators].

18.5.9.1 Activate Code

F46 + Room No. + Time (in 24-hour format: HHMM)

18.5.9.2 Deactivate Code

F45 + Room No.

Note
Only one alarm may be current for each guestroom. Any attempt to set a second alarm before the first alarm expires generates error tone.

18.5.10 Failed Wake-Up Indicator

18.5.10.1 Activate Code

F87 + User ID

18.5.10.2 Filter Interactions

Table 18 - 16 Features: Failed Wake-Up Indicator: Filter Interactions

Code	Name	Action
1	Wake-Up Failed	activate

18.5.10.3 Deactivate Code

F86 + User ID

18.5.10.4 Filter Interactions

Table 18 - 17 Features: Failed Wake-Up Indicator: Filter Interactions

Code	Name	Action
1	Wake-Up Failed	deactivate

Plexus Specification

19

19.1 Macro system

A. Power requirements

A.C. Power: 85-264VAC, Auto-ranging

Frequency: 47-63 Hz

Backup D.C. Power:24VDC, lead acid battery, 25 Ah typical

Power Consumption:150W max

Battery Draw: 6A max, 2.5A nominal

Current Rating (max):1.75 AMP AC, 7.5 AMP DC

B. Dimensions

Macro System:305H x 438W x 248D mm

C. Mounting

Wall Mount, Rack Mount, or Table Top

D. Operating environment

Temperature: 0 C - 45 C

Relative Humidity: 5 - 90%, Non-condensing

E. Wiring

2-Wire, Star Type for SL-T

4-Wire, Star Type for K-T

F. System capacities (single unit)

Macro System:

72 Ports (max with analog trunks and extensions)

112 Ports (max with 2 T1 CAS cards inserted)

124 Ports (max with 2 E1 CAS cards inserted)

G. Switching technology

Digital PCM/TDM

H. Trunk interface specifications

Signaling:	DC Loop
Dialing:	Pulse/DTMF Tone Programmable
Loop Impedance*:	600 Ohms
Polarity:	Independent
Ring:	16 - 60 Hz, 45 - 95 VAC
Loop Length:	1800 Ohms
Loop Current:	17-70 mA
Disconnect Sensing:	Loop disconnect & polarity reversal from CO switch

* = This setting is programmable through the Plexus Administration software. See chapter 6, "Software Configuration" for information.

I. Extension interface specifications

Dialing:	Pulse or DTMF Tone
Loop Impedance:	600 Ohms
Loop Length:	800 Ohms for SL-T maximum including telephone

50 Ohms for K-T maximum cable resistance

Ringer:	70 VAC 25 Hz

J. Transmission specifications

Trunk to Extension Insertion Loss: < 1 dB

Return Loss: > 20 dB

Longitudinal Balance:> 50 dB

Crosstalk Attenuation:> 65 dB

Idle Channel Noise:< -70 dBmOp

TX Reference Level:0 dBr

RX Reference Level:-3 dBr

Note
Special programmable hardware that complies with complex impedance requirements is available. Contact your dealer for more information.

19.2 Micro system

A. Power requirements

External Power Supply

A.C. Power: 110 or 220 VAC (factory set)

Frequency: 50/60 Hz

Backup D.C. Power:24VDC, lead acid battery, 10 Ah

Power Consumption:80W max

Battery Draw: 3.5A max, 1.5 nominal

Current Rating (max):1 AMP AC @ 110V, 0.5 AMP AC @ 220V, 3.0 AMP DC

B. Dimensions

Micro System: 368H x 292W x 99D mm

Micro Power Supply:254H x 150W x 90D mm

C. Mounting

Wall Mounting, Table Top

D. Operating environment

Temperature: 0 C - 45 C

Relative Humidity: 5 - 90%, Non-condensing

E. Wiring

2-Wire, Star Type for SL-T

4-Wire, Star Type for K-T

F. System capacities (single unit)

Micro System:

20.1.3 Telephone company

In the unlikely event that a Plexus system causes harm to the telephone network, the local telephone company has the right to temporarily disconnect phone service. The telephone company must provide written notice to the subscriber so that the situation may be corrected.

The local telephone company may make changes in its facilities, equipment operations, or procedures that may affect the proper functioning of a Plexus system. If they do, advanced written notification will be provided to the subscriber.

If requested, the following information must be provided to the local telephone company:

- the phone number(s) where the equipment will be connected
- the system operating mode (KF for Key mode or MF for Hybrid mode)
- the FCC registration number
- the Ringer Equivalence Number (REN)

Note
The system operating mode, FCC registration number, and REN appear on a label affixed to each Plexus system.

The REN is useful in determining how many devices may be connected to a telephone line and still have the devices ring when called. In most, but not all, areas the sum of all RENs should be 5 or less. Contact the local telephone company for verification.

20.1.4 Hearing aid compatibility

All Plexus key telephones are compatible with inductively-coupled hearing aids as prescribed by the FCC.

20.2 Part 15 of FCC rules-interference information

This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to part 15 of the FCC rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manuals, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference, in which case the user will have to correct the interference at his or her own expense.

Warning
Changes or modifications to this equipment not expressly approved by the manufacturer responsible for compliance could void the user's authority to operate the equipment.

20.2.1 Important facility installation requirements

Facility Interference codes	Loop Start: 02LS2 Direct Inward Dial (DID): 02RV2-T
Service Order code	9.0F

20.2.2 FCC Information

FCC Registration Number	4Q2USA-24094-MF.E
USO Connector	RJ11C (analog) RJ48X (digital)
Ringer Equivalence Number	0.7B

Product Warranties

21

2YHUYLHZ

The product Warranty section covers BBS Telecom's product warranty, return material authorization, and out-of-warranty services. Please refer to these sections for details on handling warranty and out-of-warranty repairs.

3URGXFW : DUUDQW\

:KDW LV &RYHUHG %\ 7KLV :DUUDQW\

BBS warrants, to the Dealer only and to no other party, that BBS Products purchased by the Dealer from BBS are free from defects in material or workmanship. The duration of this warranty is one year after the date of BBS' invoice to Dealer, unless otherwise specified in writing by BBS. If Dealer discovers within this period a defect in material or workmanship with respect to any Product, it must promptly notify BBS by calling or in writing and request a Return Authorization Number from BBS authorizing the return of the Product for warranty service. In no event shall such request be received by BBS later than thirteen (13) months from the date of invoice.

Promptly after notifying BBS, Dealer shall return the Product to BBS with a valid BBS Return Authorization Number, and with all transportation charges prepaid and in accordance with the provisions of BBS' then current RETURN AND REPAIR POLICY. Within a reasonable time after such notification and receipt of the defective Product, BBS will either repair any defect in material or workmanship in the defective Product, or will provide either new or used replacement parts, at BBS' sole option. Such repair, including both parts and labor, will be at BBS' expense, F.O.B. point of repair or replacement. BBS reserves, at its sole option, the right to replace any product or part thereof. This remedy is the Dealer's exclusive remedy for breach of warranty, and Dealer's failure to strictly comply with the terms of this shall void BBS' warranty, and BBS shall have no further obligation to Dealer or to any third party with respect to said Product.

:KDW LV 1RW &RY HUHG %\ 7KLV :DUUDQW\

BBS does not warrant (a) any Products, components or parts not sold by BBS to Dealer; (b) any BBS product subjected by Dealer or any End User to misuse, neglect, accident, improper installation, damage from lightning or any other high voltage; (c) the chance of toll fraud and the damage resulting from toll fraud; (d) damage caused by unauthorized modifications; (e) damage caused by use with equipment other than products described in BBS's official manual and service bulletins; (f) BBS Product which has been subject to repair or attempted repair by others; (g) any BBS Product where the serial number or month and year or original date of manufacture or any other means of manufacture date code has been removed, defaced or changed; or (h) any BBS Product returned not in complete accordance with the provisions of either this Agreement or BBS's RETURN AND REPAIR POICY in effect at the time.

:DUUDQW\ RI 7LWOH

In addition to the warranty set forth in the previous paragraphs, BBS warrants that it has and will convey good title to the Products free of any lien or encumbrance.

'LVFODLPHU RI :DUUDQW\

THE FOREGOING WARRANTIES ARE IN LIEU OF ALL OTHER WARRANTIES, EXPRESS OR IMPLIED, INCLUDING BUT NOT LIMITED TO THE IMPLIED WARRANTIES OF MERCHANTABILITY, FITNESS FOR A PARTICULAR PURPOSE, AND NON-INFRINGEMENT.

/LPLWDWLRQ RI 5HPHGLHV

In no case shall BBS be liable to any Dealer or any third party for any special, incidental or consequential damages based upon breach of warranty, breach of contract, negligence, strict tort, or any other legal theory. Such damages include, but are not limited to, loss of profits, indemnity, loss of savings or revenue, loss of the products or any associated equipment, cost of capital, cost of any substitute equipment, facilities or services, downtime, the claims of third parties including customers, and injury to property. BBS' liability for any damages to Dealer or to any other party will not exceed the total amount paid by Dealer under this Agreement, regardless of the form of the action, whether in contract, negligence, strict liability, tort, products liability or otherwise.

7LPH /LPLW IRU %ULQJLQJ 6XLW

Any action for breach of warranty must be commenced within fifteen (15) months following the date of invoice.

1R 2WKHU $JUH HPHQWV

Unless modified in writing signed by both parties, this Agreement is the complete and exclusive agreement between the parties, superseding all prior agreements, oral and written, and all other communications between the parties relating to the entire subject matter of this Agreement. No employee of BBS or any other party is authorized to make any warranty in addition to those made in this Agreement.

$OORFDWLRQ RI 5LVNV

This Agreement allocates the risks of product failure between BBS and Dealer. This allocation is recognized by both parties and is reflected in the price of the goods. Dealer acknowledges that it has read this Agreement, understands it, and agrees to be bound by its terms.

&KDQJH RI :DUUDQW\

BBS reserves the right to amend or modify its warranty policy, including any exclusions or limitations thereto, upon thirty (30) days prior written notice to Dealer.

5HSDLU DQG 5HSODFHPHQW :DUUDQWLHV

Any defective BBS Product repaired or replaced by BBS shall carry the longer of either the balance of the original warranty as set out herein or an additional ninety (90) day warranty under the warranty terms set out in this Agreement, said additional ninety (90) day warranty to run from the date of repair or replacement as applicable. Although BBS is not obligated to repair or replace any Product returned to BBS after that Product's warranty period has expired, in the event BBS does repair any out-of-warranty Product, the repair performed by BBS shall carry a ninety (90) day warranty from the date the repair was performed, subject to the terms and conditions set forth in this section.

5HWXUQ 0DWHULDO $XWKRUL]DWLRQ 50$ 6HUYLFH

'HDOHU 5HVSRQVLELOLWLHV

Dealer is responsible for:

- · installation of all products in compliance with BBS' installation manual, instructions and procedures
- · evaluation of the system after installation to verify it is free of programming errors
- · return of all Product in the original packaging to BBS within 14 business days after receiving a Return Authorization Number
- · back up of original configuration and programming of the system. (Any systems received by BBS will lose configuration and programming during testing procedures.)

%%6 5HVSRQVLELOLWLHV

The obligation of BBS is limited to repair or replacement (at BBS' sole option) during the warranty period. Any part, which proves to be defective in material workmanship under normal installation, use, or service, is covered under the warranty, provided the Product is returned to BBS. Transportation or shipping charges are not the responsibility of BBS.

The warranty is invalid if Product is damaged as a result of defacement, misuse, abuse, neglect, accident, destruction, alteration of the serial number, improper electrical voltages or currents, improper packaging, repair alteration or maintenance by any person other than BBS Telecom.

50$ 5HTXHVW 3URFHGXUH

Please call BBS Telecom Technical Support with RMA requests. They will help diagnose and confirm the product is not operating properly. After confirmation of improper operation, Technical Support will forward the information to Customer Support, who will assign an RMA number and call you.

2XW RI :DUUDQW\ 6HUYLFH

The Product Service Department repairs Out-of-Warranty products on a time and materials basis. Repair charges are based upon the prices in effect at the time of return. Repair charges are available upon request. An evaluation fee is applied to all out-of-warranty Products, as stipulated in the Dealer Repair Agreement Form.

The warranty on a serviced product is ninety (90) days measured from the date of service.

0LVFH00DQHRXV

To request a new RMA, please contact BBS Telecom Technical Support at:

Phone: (512) 328-9500, select "5" for technical support
Fax: (512) 328-9600
Email: bbstechs@bbstelecom.com

For all other questions regarding RMAs and Warranties, please contact BBS Telecom Customer Service at:

Phone: (512) 328-9500, select "6" for sales
Fax: (512) 328-9600
Email: bbssales@bbstelecom.com

&XVWRPHU 6HUYLFH

2YHUYLHZ

It is the policy of BBS Telecom to provide the highest quality products and services in support of our customers' needs. We recognize that we are in the business of providing not only the physical product, but also documentation, technical support, sales and marketing support, and timely product delivery. Our commitment to our customers begins with product concept and must extend long after actual product purchase and receipt.

This chapter includes information on how to reach the Sales, Ordering, and Technical Support areas of BBS Telecom.

22.2 Sales Questions

You can contact the BBS Telecommunications Pte Ltd Sales office at:

BBS Telecommunications Pte Ltd
10 Tannery Lane
BBS Building
Singapore 347773

Email: bbst_sales@bbsgroups.com.sg

Tel: (65) 6748-5600

Fax: (65) 6748-4300

22.3 Orders

To place an order or check on order status, please contact Customer Service or Sales at:

Tel: (65) 6748-5600

Fax: (65) 6748-4300

22.4 Technical Support

You can contact the BBS Telecom technical support at:

Tel: (65) 6748-5600, select menu option “5”

Fax: (65) 6748-4300

E-mail: bbst_support@bbsgroup.com.sg