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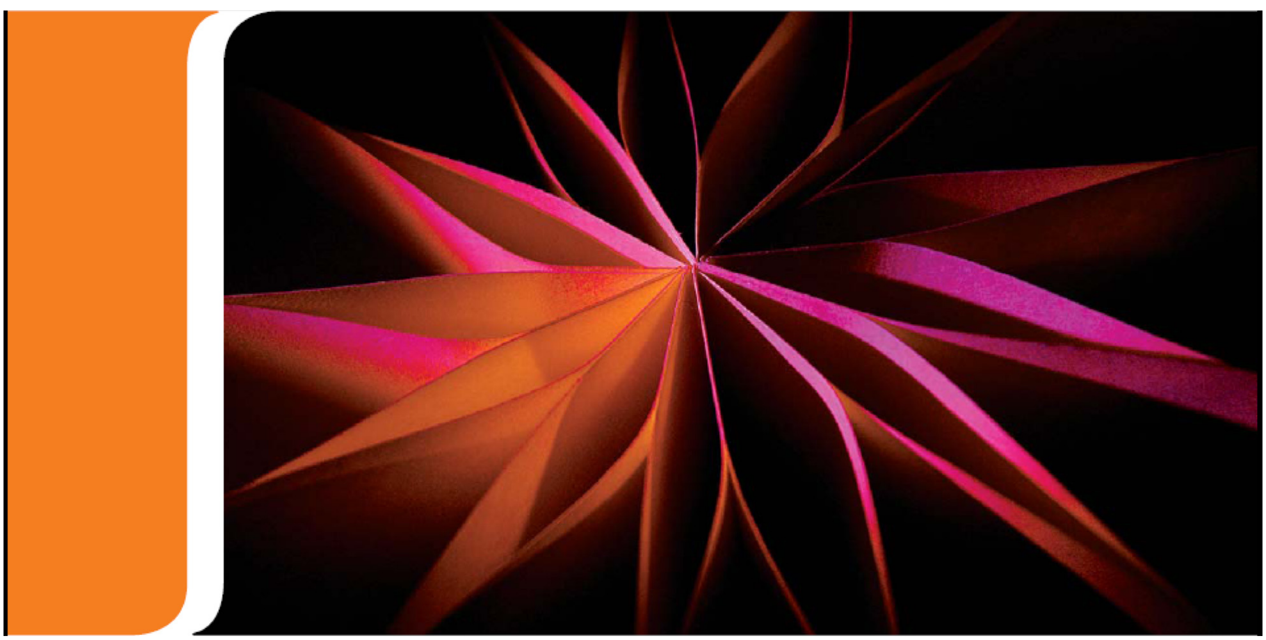
ShoreTel 12.2 System Administration Guide

Part Number 800-1630-01



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Preface

This preface provides information about the objectives, organization, and conventions of the ***ShoreTel System Administration Guide***.

Objectives of this Book

This guide explains how to use ShoreTel Director to configure, administer, and maintain the ***ShoreTel*** system.

The planning and installation procedures are described in the ***ShoreTel Planning and Installation Guide***.

Audience for this Book

This guide is written for the person who uses ShoreTel Director to administer and maintain the ***ShoreTel*** system.

Organization of this Book

This guide's organization reflects the order in which a ***ShoreTel*** system initially is configured.

The "Getting Started" section in the next chapter provides an ordered checklist to be used the first time you configure your system.

Documentation Overview

The ***ShoreTel*** system is documented as described in the following sections.

System Documentation

The following system documents are in the documentation folder on the ShoreTel DVD and can also be accessed from ShoreTel Director:

- ***ShoreTel Planning and Installation Guide*** provides information on how to plan the implementation of the ***ShoreTel*** system, as well as how to install the necessary hardware, data communications, and telecommunications elements. The ***ShoreTel Planning and Installation Guide*** can be used in conjunction with the ShoreCare ControlPoint project management tool.
- ***ShoreTel System Administration Guide*** (this guide) provides detailed information on how to administer and maintain the ***ShoreTel*** system using ShoreTel Director. This includes task-based information, as well as screen-by-screen information regarding ShoreTel Director.

Hardware Documentation

The following hardware installation documents are packaged with their associated ShoreTel voice switch or ShoreTel IP phone and appliances:

- *ShoreTel Voice Switch Quick Install Guide*
- *ShoreTel IP Phone Quick Install Guide*
- *Service Appliance 100 Planning, Installation and Administration Guide*

If the system includes the ShoreTel Enterprise Contact Center Solution, refer to the *ShoreTel Enterprise Contact Center Solution Administration Guide* and the *ShoreTel Enterprise Contact Center Solution Planning and Installation Guide*.

User Documentation

End-user documentation is installed during the ShoreTel Communicator installation. To access it, choose the **Help -> Contents and Index** menu item within the ShoreTel Communicator application.

The Telephone User Interface Analog Quick Reference and the Telephone User Interface IP Phone Quick Reference are available from the ShoreTel web site at www.shoretel.com, as well as from ShoreTel Director.

Release Notes

The *ShoreTel Server Software Release Notes* provide information about new releases and new features as well as issues that relate to new installations and upgrades. This document resides in the documentation folder on the associated DVD and can also be accessed from ShoreTel Director.

Online Knowledge Base

To access additional information or to resolve issues on ShoreTel Director, you can use the ShoreTel Technical Knowledgebase, accessible from the ShoreTel web site at www.shoretel.com.

Document Conventions

The following conventions are used in this guide:

- Data-entry fields, hypertext links, control buttons, keywords, and other items within the system management interface are in a **boldface** font.
- Information that you enter in data fields are in a data_entry font.

CHAPTER 1

Using ShoreTel Director

This chapter describes how to use ShoreTel Director. The topics in this chapter are:

1.1 Overview of ShoreTel Director

ShoreTel Director is a web-based administration and maintenance tool that lets you manage your ShoreTel system from anywhere on your IP network. The ShoreTel system has a unique distributed call control architecture as well as a suite of voice applications that provides a single image system for all users across all sites. Gone are the days of having PBXs, voice mail systems, automated attendants, or ACD systems, each with their own dedicated management interface. ShoreTel Director lets you manage, from a single web management interface, all your users and trunks, including the call control features, and all the voice applications (voice mail, automated attendant, workgroups, call detail recording, unified messaging, and desktop call control).

1.1.1 Architectural Overview

The main ShoreTel server hosts the web site for ShoreTel Director using Microsoft Internet Information Server. When you launch a web browser and navigate to the ShoreTel Director web site, the server provides HTML web pages from which you can add to, delete from, and edit the configuration of the system (see Figure 1-1). When you click **Save**, your change is sent to the server and saved in the ShoreTel database. All other system components are automatically and immediately notified and updated.

In addition to the configuration panels, ShoreTel Director has a maintenance interface for the ShoreTel system. When you navigate to a maintenance panel in ShoreTel Director, system status is displayed, and you can issue maintenance commands. The commands are passed to and executed by the server. If you have distributed ShoreTel application servers, you can navigate from the main server to each distributed server through ShoreTel Director to view status and issue commands to the distributed server.

ShoreTel Director is supported by the following:

- MS Internet Explorer 8.0 - ShoreTel Director and ShoreTel Communicator for Web
- Safari 4.0 - (On Apple Corporation's Macintosh computers)
- Firefox 3.6 - (On Microsoft Corporation's Windows-based computers and Mac OS)

If you are running ShoreTel Director under a Windows operating system (such as XP SP2), the Director site (URL) must be added to Internet Explorer's list of trusted sites.

1.1.2 Multi-level Management

The ShoreTel system provides in-depth access levels to ShoreTel Director. System parameters for administrative permissions allow many administrative roles to be defined so as to provide only as much access to the system as each user requires. By default, the initial system administrator has access to everything on the system. However, by using the administrative permissions pages, you can define site administrators, directory list managers, read-only users, and more. Each user who needs to access Director can be assigned a level of permission tailored for his needs.

1.1.3 Multi-user Management

ShoreTel Director allows simultaneous access to ShoreTel Director by multiple users. To ensure data integrity, the database is locked during save transactions in ShoreTel Director. If another user tries to save changes while the database is locked, ShoreTel Director advises the user that the changes were not saved; the user simply needs to save the changes again.

Most changes to the database are completed within one second, so the probability of attempting to save while the database is locked is low.

If two users open the same record at the same time, the user who saves last "wins," since changes are processed in serial order. If two users open the same record at the same time and the first user deletes the record, the second user receives an error message upon trying to save.

1.1.4 External Telephone Number Formatting

Unlike the ShoreTel Communicator, which has a location as a reference point, ShoreTel Director is global and has no inherent location; it has no inherent relationship to local exchanges or countries. The following is a list of number entry rules:

All external numbers in ShoreTel Director – must be entered in canonical format, as follows:

- +C… (A…) S…

  "+" = International designation
  "C" = Country code
  "A" = Area code (also known as a city code)
  "S" = Subscriber number

DID Numbers – Must be entered in canonical format.

- +1 (408) 331-3300 (U.S., Canada)

Message Notification Destinations – Must be entered in canonical format and *must not* include a trunk access code. The number is presented back to the user in canonical format.

- +1 (408) 331-3300 (U.S., Canada)

System Directory Entries – Must be entered in canonical format. The number is presented back to the user in canonical format.

- +1 (408) 331-3300 (U.S., Canada)

Call Forward Destinations – Must be entered in canonical format and *must* include a trunk access code. The number is presented back to the user in canonical format with the trunk access code in front of the number.

- 9+1 (408) 331-3300 (U.S., Canada)

Off-system extensions can be used as call forward destinations. You must enter the off-system extension without a trunk access code (for example, 8 or 9) for proper operation.

Table 1-1 gives examples for all the countries supported by ShoreTel Director.

Table 1-1 International Phone Number Examples

Phone Number	Country
+1(408) 331-3300	U.S., Canada
+31 70 348 6486	Netherlands
+33 8 36 68 31 12	France
+34 91 845 6078	Spain
+44 20 7634 8700	UK
+49 69 571903	Germany
+55 61 429 7777	Brazil
+60 3 2693 5188	Malaysia
+61 2 9360 1111	Australia
+65 736 6622	Singapore
+852 2508 1234	Hong Kong

1.1.5 Preferences

ShoreTel Director saves certain preferences in cookies on the client PC for ease of use. The Preference page shown in Figure 1-1 lets you configure some of these preferences, including the way to record auto-attendant prompts and certain default settings. To view this page, click Preferences under the Administration link in the ShoreTel Director application (described later in this chapter).

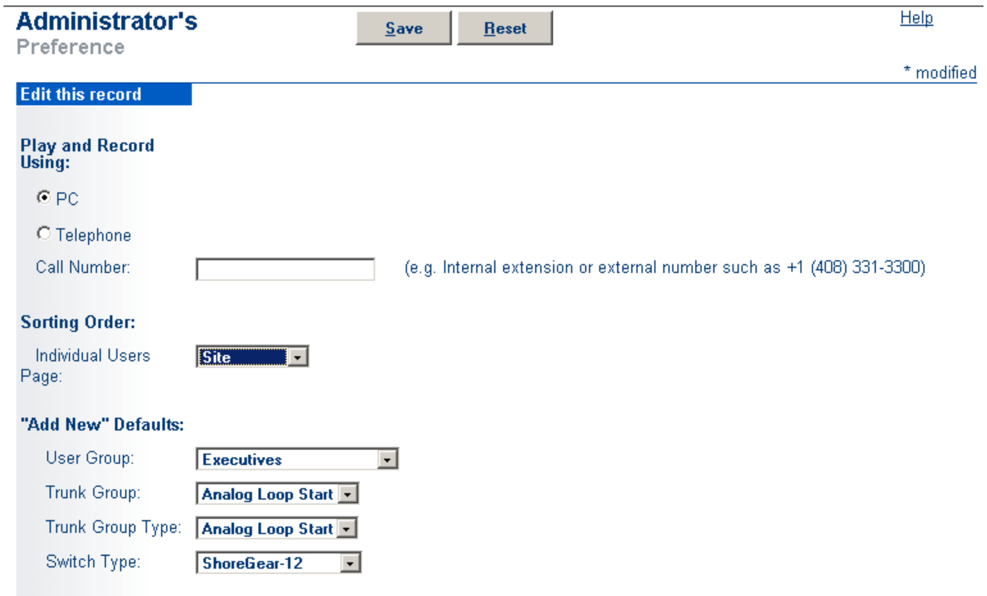


Figure 1-1 Preference Edit Page

The selectable parameters in the Administrator's Preference window are as follows:

- **Play and Record Using:** This area lets you specify one of the following methods for recording and playing an auto-attendant prompt.
  - — *PC* – This selection is for recording and playing a greeting through a microphone and PC loudspeaker. A sound card must exist in the computer for this option.
  - — **Telephone** – Select to play and record by use of a telephone handset. For this option, the Call Number field must have the telephone number or extension that you want the system to use to call you.
- **Sorting Order:** Lets you specify the default column used to sort users on the Individual Users page.
- **"Add New" Defaults:** This section lets you set default values that are used throughout the system for creating new profiles. The parameters for which you can set values include:
  - — User Group
  - — Trunk Group
  - — Trunk Group Type
  - — Switch Type

  For each parameter, click the field and select the value that you want to use from the drop-down menu.

In addition to the preferences that are explicitly exposed, ShoreTel Director keeps other preferences as cookies and uses them to predict good defaults when it adds new records. The following information is also stored locally in cookies:

- Extension numbers
- DID numbers
- Sort order for every list page

1.1.6 Prompts

Prompts on the ShoreTel system can be imported into the system by using µ-law, WAV file format (CCITT µ-law, 8KHz, 8 bit, mono). If you want your prompts to match the voice of the ShoreTel system, contact Worldly Voices at the web site www.worldlyvoices.com. Request that “Connie” record your prompts. Worldly Voices provides this service for a nominal fee and provides rapid turnaround.

1.2 Starting ShoreTel Director

ShoreTel Director is a web application hosted on a ShoreTel server and accessed over the network. Before starting ShoreTel Director, you must have obtained the ShoreTel Director Uniform Resource Locator (URL), your user ID, and password from a system administrator.

Browser Requirements for ShoreTel 12, 12.1, and 12.2

- Microsoft Internet Exporer (IE) 8 or later.

Browser Requirements for ShoreTel 11, 11.1 and 11.2

- MS Internet Explorer 7.0 - ShoreWare Director
- MS Internet Explorer 8.0 - ShoreWare Director and ShoreTel Communicator for Web
- Safari 4.0 - (On Apple Corproation’s Macintosh computers)
- Firefox 3.6 - (On Microsoft Corporation’s Windows-based computers and Mac OS)

To start ShoreTel Director, do the following:

Step 1 Launch a browser.

Step 2 In the URL field, enter the following:

http://*<ShoreTel server name> | <IP address>***/ShoreWareDirector**

Click **Go** or press **Enter**. The login screen appears as shown in Figure 1-2. Also, the build number for the ShoreTel software appears above the copyright number in the lower-left corner of the window.

Figure 1-2 Login Page

Step 3 In the User ID field, type your user ID or the default User ID (“admin”).

Step 4 In the Password field, type your password or the default password (“changeme”).

You must create a user with administrator privilege to gain full access to ShoreTel Director features. The admin and changeme defaults must be changed after you assign full System Administrator permission to a user. See the "Administrative Permissions" on page 59 for additional information and for information about giving users administrative permission.

Step 5 Click **Login.** Upon first-time login to a new system, the ShoreTel Director Welcome screen appears as shown in Figure 1-3. Upon subsequent logins, unregistered systems skip the Welcome screen and display the registration window shown in Figure 1-4.

Figure 1-3 Product Registration Page

1.3 Product Registration

After you install or upgrade to ShoreTel 12.2 and launch ShoreTel Director for the first time, a Welcome screen requests you to register the ShoreTel software and requests your ShoreTel software system keys. It provides you with the following registration options:

ShoreTel encourages you to register ShoreTel Director promptly so that we have the most up-to-date information concerning your ShoreTel products and installation.

NOTE For continued use of ShoreTel Director, the system administrator must register ShoreTel software within 45 days. You can request a new license during registration.

To register ShoreTel Director, take one of the following actions:

- Select **Now** to register immediately. ShoreTel Director opens the **Contact Information** page for entering registration information, as described in the "Contact Information" on page 24.
- Select **Later** to register at another convenient time. With this selection, ShoreTel Director opens at the **Quick Look** page.

The registration data goes to ShoreTel over a secure connection to ensure integrity and privacy. For systems that are not online, an email transmission option is available

If an installation does not have adequate or current licenses, ShoreTel Director opens at the **License Preview** page when you have completed or skipped registration. See the "Requesting the ShoreTel Software License Key" on page 25 for information.

1.3.1 Contact Information

When you select the **Now** option from the Welcome screen, the ShoreTel software registration starts. The Contact Information screen, appears as shown in Figure 1-4.

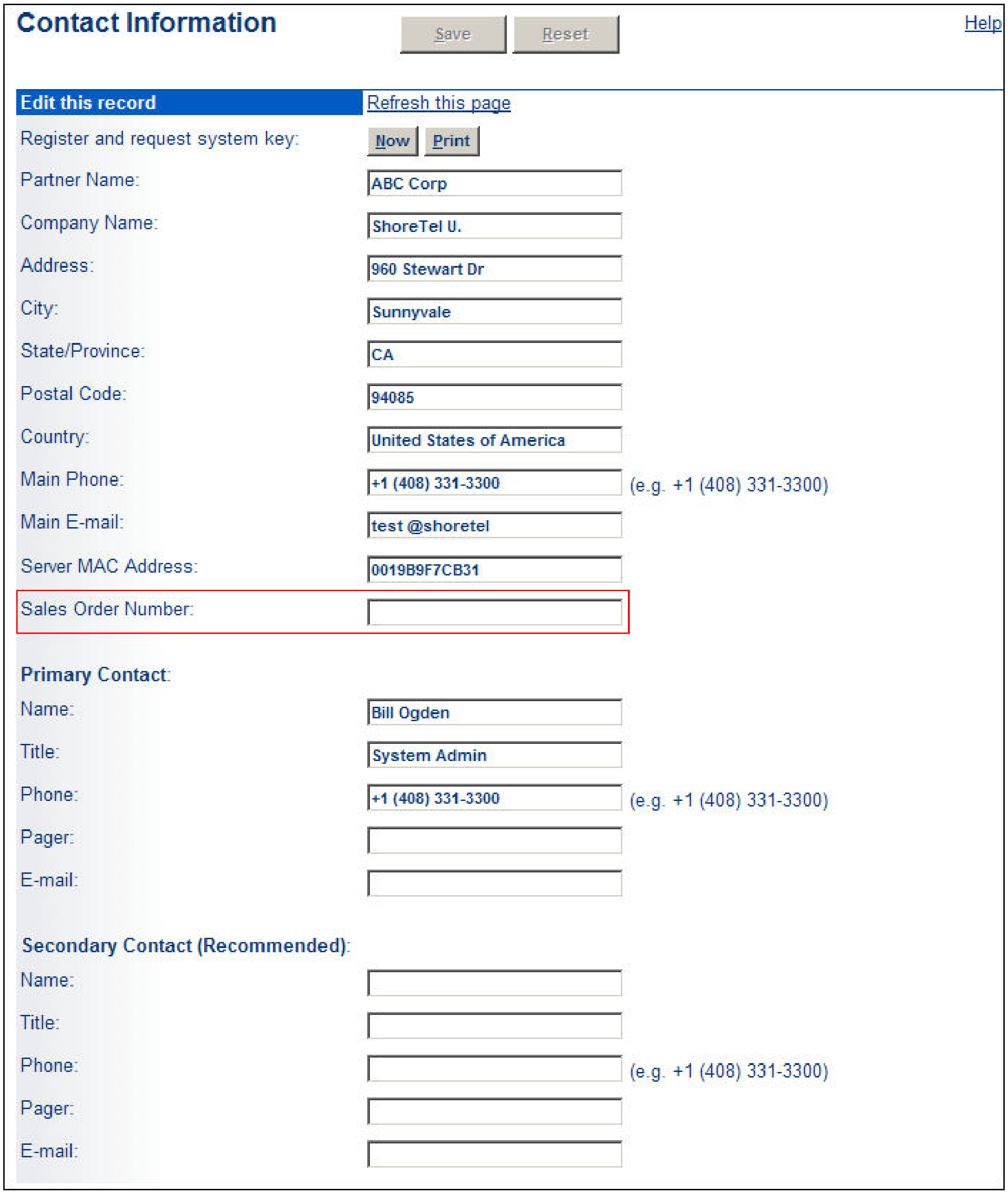


Figure 1-4 User Contact Information

To specify contact information:

Step 1 Type the requested information in the applicable fields.

The Server MAC Address field is automatically populated with information from the ShoreTel Server. You should change this information only if you want a license for a server other than the one to which you are currently connected. If you have changed this information but instead want the defaults, click **Refresh this page**.

The Sales Order Number is on the ShoreTel packing slip. (Supplying this information is optional for system upgrades.)

Step 2 Click **Save**.

Step 3 Click **Now** to register immediately.

The contact information goes to ShoreTel, and the License Preview page appears. See the "Requesting the ShoreTel Software License Key" on page 25 for information about requesting the ShoreTel license key.

Step 4 Click **Print** to print out and send in the contact information at a later time.

The contact information is displayed. Save this information as a text file. Later, when you are ready to complete the registration process, send the file to shorecare_admin@shoretel.com.

1.3.2 Requesting the ShoreTel Software License Key

After completing and registering the Contact Information form, the License Preview page appears as shown in Figure 1-5. You must request a license key.

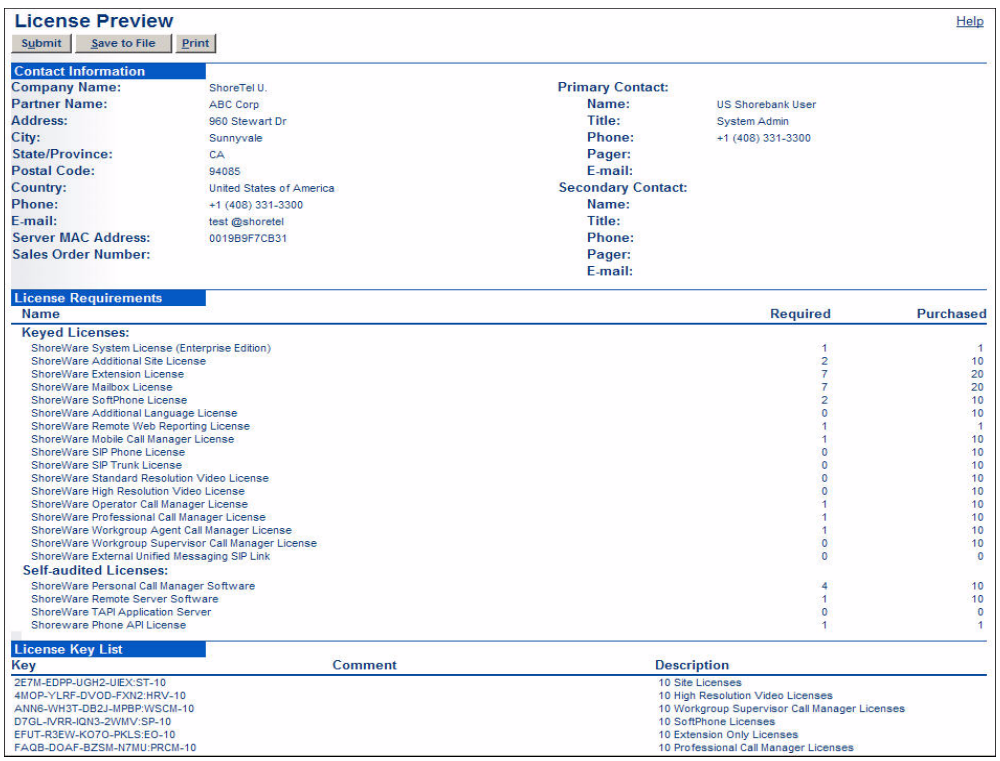


Figure 1-5 License Preview Page

To request a license key, do the following:

Step 1 Review the information in the License Preview page.

Step 2 Do any of the following:

- Click the Print button at the top of the page to print the information.
- Click the **Submit** button to send the request immediately to ShoreTel, Inc. After verifying the information, ShoreTel emails the license key within three business days.
- Click the **Save to File** button to save the request for later submission.

After you have registered ShoreTel Director, the Quick Look page appears shown in Figure 1-6.

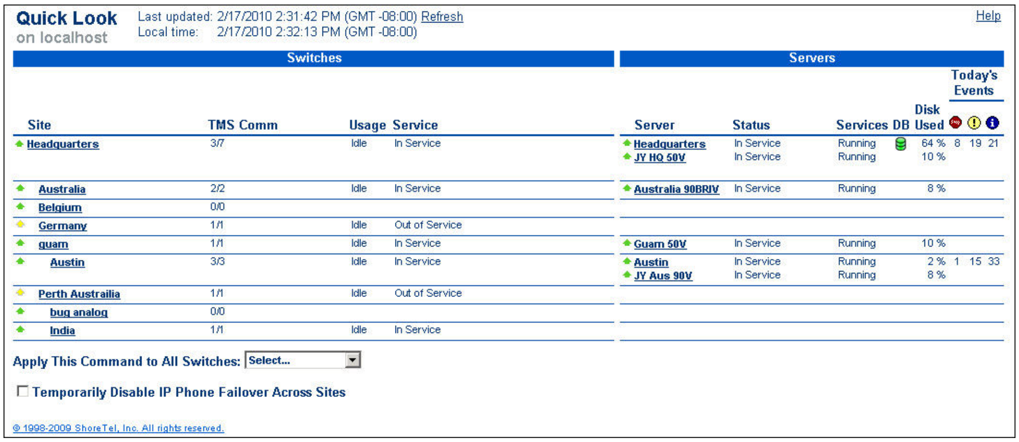


Figure 1-6 Quick Look Page

You can use ShoreTel Director to configure and monitor a ShoreTel system. We recommend you first create a unique user with system administration privileges and a unique password, and then change the default login and password. See the "Administrative Permissions" on page 59 for information about creating users and assigning administration permission.ShoreTel Director sessions terminate after 60 minutes of inactivity.

Until the license key arrives, you can click on Later in the ShoreTel Director Welcome screen to enter ShoreTel Director. You have up to 45 days to install the license key.

NOTE Installation of any release after ShoreTel 12 prompts for re-registration.

1.3.3 Installing the License

When you receive your licence keys, you must install them in ShoreTel Director. To install the license, do the following:

Step 1 View the license packet that you received from ShoreTel.

Step 2 Launch ShoreTel Director.

Step 3 In the ShoreTel Director menu click **Administration > System Parameters >Licenses > Keys**. The License Key page appears.

Step 4 Click the **New** button at the top of the page. The License Key Info dialog box appears.

Step 5 In the **Key** field, enter the license key that you received from ShoreTel.

Step 6 In the **Comment** field, enter a description of the license.

Step 7 Click **Save**.

Step 8 Repeat Steps 4 through 7 for each license key that you have to install.

1.4 Director Components

ShoreTel Director is the administration interface for the ShoreTel system. This section provides a brief discussion about how to navigate and use ShoreTel Director. To access ShoreTel Director, do the following:

Step 1 Launch a browser. ShoreTel Director supports Microsoft Internet Explorer (IE) 8 or later.

Step 2 In the URL field, enter the following:

`http://<ShoreTel server name or IP address>/`**`ShoreWareDirector`**

Click **Go** or press **Enter**. If you have installed a license key, the ShoreTel Director interface appears as shown in Figure 1-7. If the registration page appears, click Later to move on.

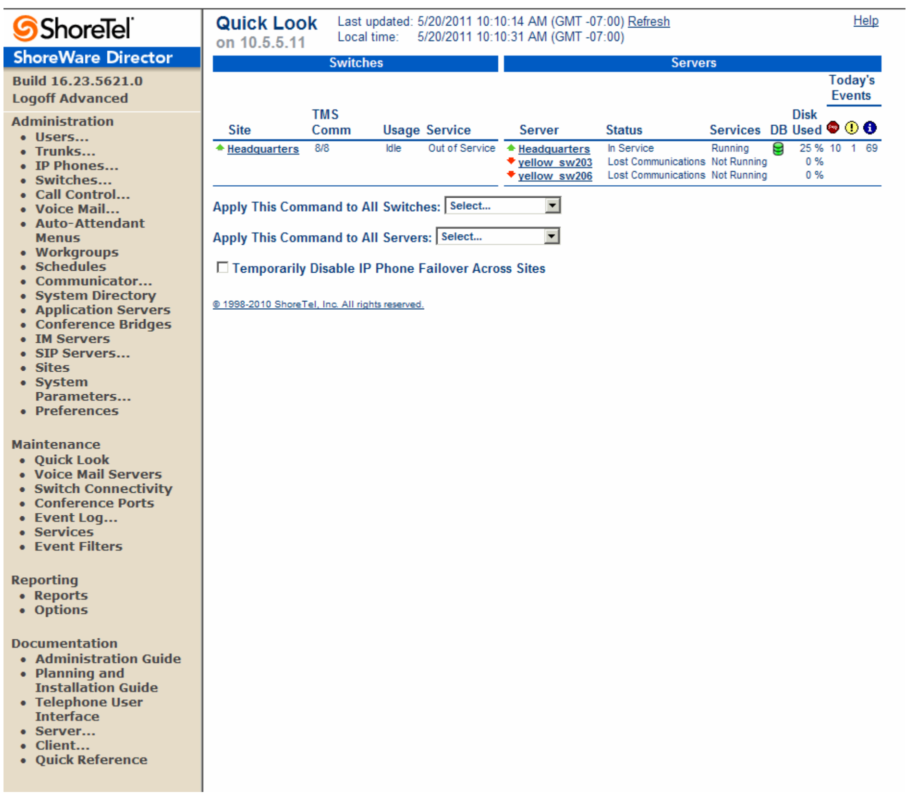


Figure 1-7 ShoreTel Director Interface

The interface includes two columns. The column on the left is the Navigation Frame. The column on the right is the Data page. Both are discussed below.

1.4.1 Navigation Frame

The navigation frame is located on the left side of the Director page and provides access to the following links:

- **Build**: Indicates the version of the ShoreTel system that you are running.
- **Logoff**: Lets you log out of the ShoreTel Director as the current user.

  NOTE You are automatically logged off after 60 minutes of inactivity unless you are viewing the Quick Look page or a Switch maintenance page.
- **Administration**: Lets you configure the ShoreTel system.
- **Maintenance**: Lets you view status information about the components installed in your system.
- **Reporting**: Lets you run ShoreTel reports.
- **Documentation**: Lets you download ShoreTel product documentation.

1.4.2 Data Pages

Data pages display on the right side of the ShoreTel Director page. These pages let you add, delete, or edit system configuration parameters, view status, and issue commands. These pages are designed as list pages, edit pages, and maintenance pages.

1.4.2.1 List Pages

List pages identify objects that are created in the system in the category that you have selected. These pages generally provide categorical information about the objects.

Adding to Objects

New, **Add new**, and **Go** links and **New** buttons allow you to add objects to the system configuration. Clicking the prompt may display a configuration page or a dialog box.

Editing Objects

To display for viewing or editing the profile of an object in a List page, click on the object. Principal list objects are shown in bold characters. Values used to define the principal object that are themselves configurable are underlined. You can click such values to view and edit their parameters.

Sorting

The records on each list page are presented in a default sort order. Most pages allow you to change the sort order by clicking a column heading. Elements in the column are rearranged in ascending order only.

List Paging

To assist enterprises with large amounts of data, several configuration screens offer searching and sorting of the data used in a pertinent field. For example, when selecting members of a hunt group, you can search for last names, extensions, and more. If more data is returned than fits the window, you can page up and down through the results. Hunt groups, extension lists, workgroup agents, and call handing delegation offer paging.

1.4.2.2 Edit Pages

Edit pages let you to view and edit object profiles. Edit pages have up to five control buttons at the top of the page. Table 1-2 on page 30 provides information about the buttons. You must enter a value in any field to activate these buttons. Some edit pages also have a tab bar as shown in the Figure 1-8. The tabs open additional configuration pages for the object. You must click the Save button to save your changes.

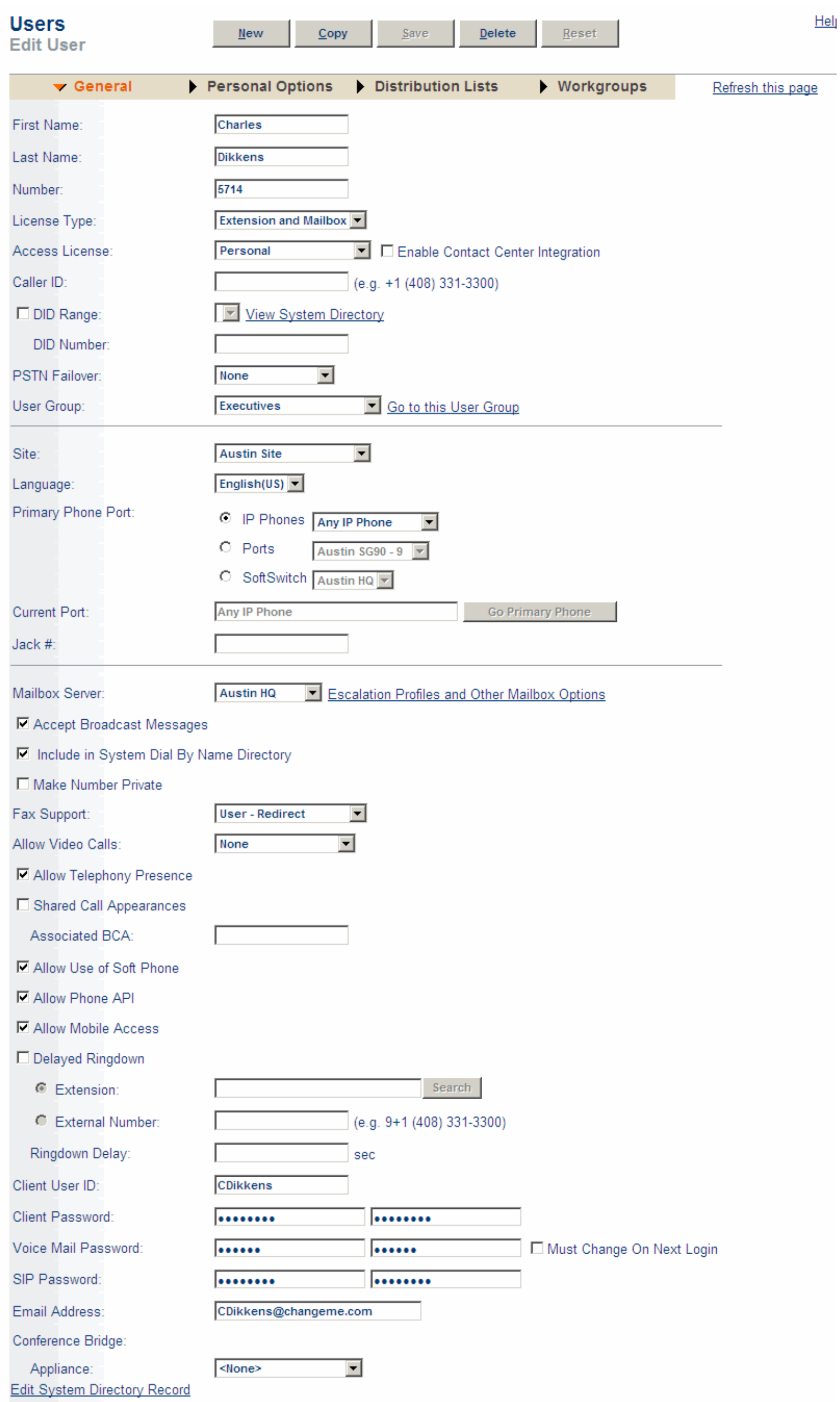


Figure 1-8 Edit User Page

Table 1-2 Control Buttons

Button	Function
New	Creates a new object profile by using default values.
Copy	Create a copy of the current object profile that you can use to create a new object profile. Some values, such as extension numbers, are automatically generated for the new profile.
Save	Saves the changes you make to the profile.
:Delete	Removes the current profile from the system.
Reset	Reverts to the last saved profile.

NOTE Individual data-entry fields, drop-down lists, and option buttons are described in the appropriate sections throughout each chapter.

ShoreTel Director saves certain parameters on your computer locally (using cookies) to help present the most relevant default configuration when you click New or Copy. This includes extension numbers and DID numbers.

1.4.3 Getting Help

To get help that pertains to the page you are working on, you can press the ***F1*** key or click the *Help* link in the data-entry frame, which provides access to the relevant chapter from this guide. From the navigation frame, you can access this complete guide, or other system documentation, by clicking the Documentation link to expand the list of guides. The following documentation is provided:

- ***ShoreTel 12: System Administration Guide***
- ***ShoreTel Planning and Installation Guide***
- Telephone User Interface Guide IP Phone
- Telephone User Interface Guide Analog Phone
- Server Release Notes
- Client Release Notes

Clicking any of the titles invokes a Portable Document Format (PDF) copy of the corresponding document.

1.5 Getting Started

This section lists a summary of tasks to perform to configure your system for the first time. This ordered list also follows the order in which this guide is organized. Before you begin configuring the system, make sure it has been properly installed as described in the ***ShoreTel Planning and Installation Guide***.

- ❑ Launch ShoreTel Director. See the "Starting ShoreTel Director" on page 21 for information about launching ShoreTel Director.
- ❑ Register the ShoreTel software and request a license key or keys. See "Requesting the ShoreTel Software License Key" on page 25 for information about registering ShoreTel software and requesting license keys.

We encourage prompt register ShoreTel Director so that we have the current information about your ShoreTel products and installation.

❑ Install your license key or license keys if you have them. You have up to 45 days to install the licenses after which you cannot continue to use the ShoreTel software. See the "Installing the License" on page 26 for information about installing license keys.

Until you have updated all required licenses, ShoreTel Director will continue to open to the License Requirements page after login.

❑ Configure system parameters using the System Parameters link:

- Install any new licenses your installation requires from the Licenses page.
- Specify the dialing conventions to use throughout the system in the Dialing Plan page. See the "Setting Dial Plan Parameters" on page 35 for more information about dialing plans. The dialing conventions include extension length as well as the dialing plan reservations for extensions and trunk access codes.
- Configure the system's extensions from the System Extensions page. See the "System Extensions" on page 41 for more information about system extensions. Review the default system extensions and change the defaults if you need to use these extensions for other purposes.
- Specify the languages you want to make available for the system. (Be sure you have appropriate licenses for the languages.)
- Review password and log file settings on the Other page. See the "Other" on page 45 for more information about other configurable parameters. You should be able to accept all the defaults.

❑ Create and configure the sites that you want your ShoreTel system to have using the Sites navigation link. Chapter 3: ShoreTel Sites on page 65.

- Review the following default Headquarters site parameters. Click the Headquarters link on the Site list page.

  — Country location for the site
  — Local area codes for 7- and 10-digit dialing
  — Emergency call back number (if using ISDN PRI)
  — Time zone (for correct date and time for caller ID telephones)
  — Admission control bandwidth (for multiple-site configurations)

  You cannot configure Night Bell Switch or Paging Extension until the proper switch is configured, nor can you configure Operator Extension or Fax Redirect Extension until the proper users are configured. You will need to return to this page later.

- Set the IP address range for the IP phones at any remote sites. You define IP address ranges so that IP phones are assigned to the correct site. IP phones not assigned to a remote site are associated with Headquarters.
- Configure additional sites if desired.

❑ Configure additional ShoreTel servers by using the Application Servers page. See Chapter 4: Configuring Application Servers on page 73.

- Name the server and assign it to a site.
- Create the ShoreTel server.

- Set the voice mail and auto-attendant extensions.
- Assign a user group to the server.

❑ If you are using the ShoreTel Enterprise Contact Center Solution, you need to configure it. Refer to the *ShoreTel Enterprise Contact Center Administration Guide* and the *ShoreTel Enterprise Contact Center Installation Guide* for more information.

❑ Configure ShoreTel voice switches using the Switches page. "Configuring Switches" on page -89 for more information.

- Select the role that you want to switch to perform for the site.
- Identify the site where you want to use the switch.
- Select the switch model you want to use.
- Create the switch profile.
- Provide a name and description and use the Find Switches button to discover each voice switch on the network.
- Specify the ShoreTel server that you want to manage the switch.

Your ShoreTel voice switches must have a valid IP address from a DHCP server or from the BOOTP server on the ShoreTel server, or an address statically configured from the maintenance port (24, T1, and E1 only).

❑ Configure IP phones using the IP Phones link. See Chapter 8: Configuring IP Phones on page 191 for more information.

- Add IP phone ports to ShoreTel voice switch-120/24, ShoreTel voice switch 90, ShoreTel voice switch-60/12, ShoreTel voice switch 50, ShoreTel voice switch-40/8, ShoreTel voice switch 220T1/T1A/E1 and ShoreTel voice switch-T1/E1 voice switches supporting IP phones. For each port you assign to IP phones, the switch supports five IP phones. For more information, Chapter 8: Configuring IP Phones on page 191.
- Set the boot parameters of the IP phones. ShoreTel IP phones are set to find boot information from a DHCP server. If your installation has other requirements, use the IP phone set-up menu to set server and boot configuration parameters. For more information, see the ***ShoreTel Planning and Installation Guide***.

You can speed up the installation by using the Extension Assignment feature. See the "Using Extension Assignment" on page 410 for information.

❑ Configure the following users before you add general users to the system:

- During installation, a system administrator is set up. Assign a person at your site to this role. When you assign a system administrator, the default user ID and password must be changed. Make a note of the new user ID and password, since the defaults (admin and changeme) will no longer be available.
- Configure an operator for each site. See the "Administrative Permissions" on page 59 for more information.

  This is the extension reached when 0 is dialed from the telephone. Note that operators can span sites.

  Configure a "user" as the Fax Redirect extension for each site.

- Configure a user as the default Personal Assistant for all other users. This is the user that calling parties are routed to when they dial "0" in a user's mailbox. It is important that you configure the default Personal Assistant before adding the bulk of the users so that appropriate defaults can be assigned. If you omit this step, you may have to spend time reconfiguring the users later.
- Configure the Call Handling Modes Defaults and assign the Personal Assistant.

❑ Complete configuration of sites:

- Return to each site and complete the configuration for Night Bell, Paging, Operator, and Fax Redirect.
- If you have added additional servers, return to each **Site** edit page and reconfigure as appropriate.

❑ Configure all trunk groups and trunks:

- Configure trunk groups from the Trunk Groups page. You can modify the default trunk groups, add new trunk groups, and assign individual trunks for Make Me conferencing.
- Configure the trunks from the Individual Trunks page. For the ShoreTel voice switch-T1, return to the Switch edit page and assign all the channels on the T1 to the proper trunk group.

❑ Configure the users from the Users pages:

- Configure the user groups, including all the Class of Service (CoS) permissions from the User Group edit page. You can modify the default user groups or add new trunk groups. Be sure to grant access to any new trunk groups you have added.
- Configure all the users from the Individual Users edit page.
- Configure any anonymous telephones from the Anonymous Telephone edit page.

❑ Configure call control parameters from the Call Control link. Set up hunt groups and paging groups, as needed.

❑ Configure voicemail parameters and system distribution lists from the Voice Mail Options and System Distribution Lists pages.

❑ Configure the auto-attendant parameters from the Auto-Attendant edit pages:

- Configure each auto-attendant menu from the Menus page.

❑ Set schedules from the *Schedules* link. These may be used by the auto-attendant or paging groups.

- Configure the workgroups from the Workgroups edit page:
- Configure the workgroup.
- Configure the call handling modes.
- Configure the queue, including prompts.

❑ Configure the system directory from the System Directory edit page:

If you use Microsoft Exchange and Microsoft Outlook, you can leverage Contacts on the Exchange Server for common contact information.

CHAPTER 2

Setting Up System Parameters

This chapter describes how to set system-wide parameters by using ShoreTel Director. The topic sections in this chapter are as follows:

2.1 Setting Dial Plan Parameters

The dial plan defines the numbering convention your ShoreTel system uses to route calls. The system uses the dial plan to parse dialed numbers—whether from internal users or the Public Switched Telephone Network (PSTN)—and to direct calls appropriately. The dial plan can include extensions, site codes (pre-extensions), access codes for trunks, and permission codes.

This section describes how to set the parameters used to create number strings in your dial plan. These parameters are set using the Dial Plan page in ShoreTel Director. This page allows you to do the following:

- Specify the lead digit used in a string.
- Specify the number of digits included in a string.

  NOTE You cannot reduce the number of digits included in an extension after the parameter is set.

- Specify how the sting is used in the system.

Before you begin, review your ShoreTel system deployment and topology, and the local Telco dial plan and dial rules for each ShoreTel site that you deploy.

To set the string parameters used in your dial plan, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > System Parameters > Dial Pla**n. The Edit Dialing Plan page appears as shown in Figure 2-1. See Table 2-1 on page 37 for information about the configuration elements that appear on the page.

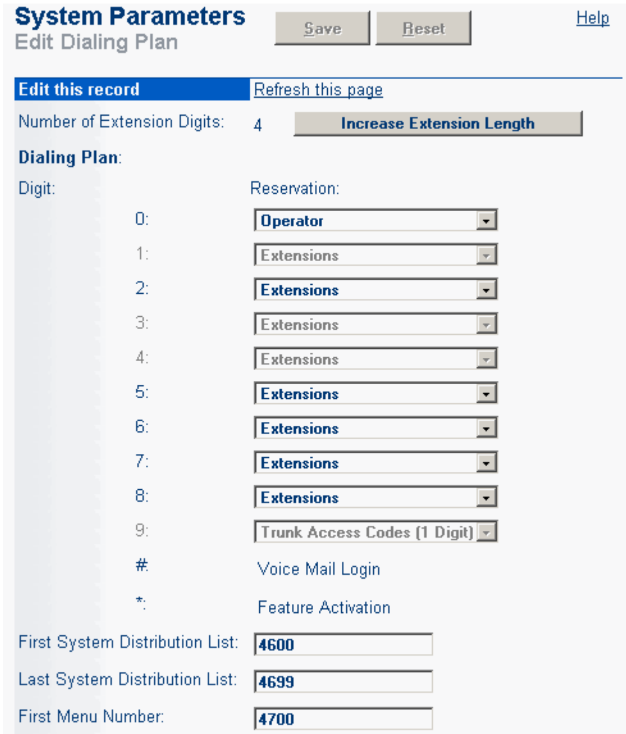


Figure 2-1 Edit Dialing Plan Page

Step 3 In the Digit column, identify the number or character that you want to use for the leading digit in this string.

> **NOTE** All available digits are preconfigured as lead dial strings. Reconfigure only those dial strings that you want to use for special purposes.

Step 4 In the **Reservation** field next to the digit, select the parameter that you want to use for this digit string. See Table 2-2 on page 37 for information about the available parameters.

IMPORTANT Extensions must not conflict with the leading digits of emergency telephone numbers, since the ShoreTel system allows users to dial emergency numbers with or without a trunk access code. If you are deploying a global voice network, this must be considered for all emergency numbers.

Step 5 Repeat Step 3 and Step 4 for all of the digits that you want to change the configuration.

Step 6 Click **Save** to save your changes.

IMPORTANT Once you set and save the Reservation parameter, it cannot be changed. The field is unselectable after the change is saved.

Table 2-1 System Parameters Page: Dial Plan Elements

Element	Description
Number of Extension Digits	Shows the number of digits currently uses in ShoreTel extensions. The default is 3 digits.
Increase Extension Length	Allows you to increase the number of digits used in ShoreTel extensions.
Dialing Plan	This section allows you to set parameters used for extension numbers. You must specify how you want the ShoreTel system to interpret each dialable, leading digit.
Digit	The numeric and ASCII characters used as leading digits.
Reservation	Allows you to set the extension parameters that you want to use with each leading digit. See Table 2-2 for information about extension options.

Table 2-2 Reservation Options

Option	Description
Extension Prefix (n digits)	Lets you specify the number of digit used in extension prefixes that have this leading digit. Extension prefixes can be up to 7 digits in length. **NOTE** The Configure Extension Prefix Warning window appears with a list of each of the sites in your system. Next to the list of sites you will find a blank field that requires you to enter the desired extension prefix. This prefix will be appended to every dialed number at that particular site. Make sure to back up the system before clicking Save.
Extensions	Reserve this digit as the leading digit in an extension. **NOTE** The digit “0” cannot be reserved as the lead digit in extensions.
Not Used	Does not allow this digit to be used as a lead digit.
Operator	Reserve this digit for use as the extension used to access the ShoreTel operator. The default value is zero (0). In international applications, zero is often used as the access code for trunks. This sets a potential for conflict. ShoreTel recommends that international customers standardize globally on a single trunk access code for the purposes of network call routing (for example, use “9” for all trunk groups)

Table 2-2 Reservation Options (Continued)

Option	Description
Trunk Access Codes [1 Digit]	Reserves this digit for use as a trunk access code. **NOTE** When you make a number the leading digit in a trunk access code, using the same number as the leading digit in extensions can cause the system to misroute calls.
Trunk Access Codes [2 Digit]	Reserves this digit as the lead digit in two-digit trunk access codes. **NOTE** When you make a number the leading digit in a trunk access code, using the same number as the leading digit in extensions can cause the system to misroute calls.
Trunk Access Codes [3 Digit]	Reserves this digit as the lead digit in three-digit trunk access codes. **NOTE** When you make a number the leading digit in a trunk access code, using the same number as the leading digit in extensions can cause the system to misroute calls.

2.1.1 Increasing the Extension Length

The ShoreTel system lets you increase the number of digits the systems use for phone extensions from the default number of three digits to five digits. To match the new number when you increase the number of extension digits, you must also add one or more numbers to the beginning of that extension for existing numbers, including mailboxes, menus, and distribution lists. Be sure that the added number or numbers do not conflict with other access codes in the system's dial plan.

To increase the number of digits for phone extensions, do the following:

Step 1 Launch ShoreTel Director.

Step 2 In the ShoreTel Director menu, click **Administration >System Parameters > Dial Plan**. The System Parameters: Edit Dial Plan page appears.

Step 3 Click the **Increase Extension Length** button. The Increase Extension Length Warning dialog appears as shown in Figure 2-2.

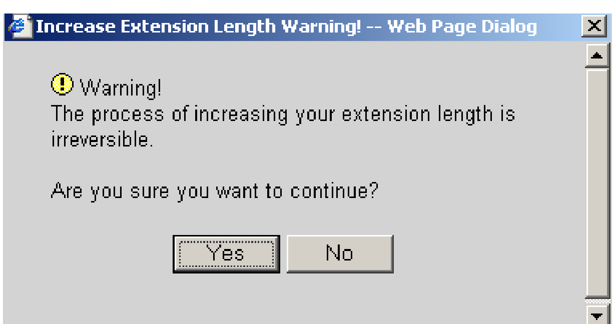


Figure 2-2 Warning Dialog Box

Step 4 Click **Yes** to increase the extension length. The Increase Extension Length dialog appears as shown in Figure 2-3

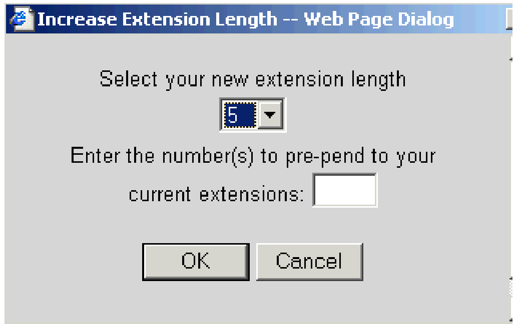


Figure 2-3 Specifying New Extension Digits

Step 5 In the **Select your new extension length** field, select the number of digits that you want to use for ShoreTel extensions.

Step 6 In the **Enter the number(s) to pre-pend to your current extension** field, enter the number or numbers that you want to appear at the beginning of extensions.

IMPORTANT Make sure that the numbers that you pre-pend to the extension are not used elsewhere in your dial plan in such a way that they conflict with other strings such as trunk access codes, the operator extension, and emergency numbers.

Step 7 Click **OK** to apply the change.

2.2 Digit Translation Tables

Digit translation tables let you convert numbers from other dial plans to ShoreTel numbers.

When designing and creating a ShoreTel system, you might plan to integrate the system with another phone system that uses a different dial plan. To make the transition between systems invisible to users, you can create digital translation tables in ShoreTel Director. These tables allow you to compensate for a difference in the number of digits in each plan. The ShoreTel system applies the adjustment when it passes calls between the two systems. You can then associate the digital translation table with a trunk bridging systems or the Simplified Message Desk Interface (SMDI) module in an application server allowing users to access their legacy voice mailbox. In either case, users do not have to change their dialing habits.

This section describes how to create digit translation tables. Information about applying digital translation tables appears in the sections about trunks and application servers.

Digital translation tables let you adjust the extension format used in your ShoreTel dial plan to the format used in another dial plan of another phone system. You can specify the number of digits and the lead digit used by each system. When applied, the ShoreTel system adds or deletes the digits you specify depending on the direction of the call route.

2.2.1 Creating Digit Translation Tables

To create a digit translation table in ShoreWare, do the following:

Step 1 Launch ShoreTel Director.

Step 2 In the ShoreTel Director menu, click **Administration > System Parameters > Digital Translation Tables**. The Digital Translation List page appears as shown in Figure 2-4.

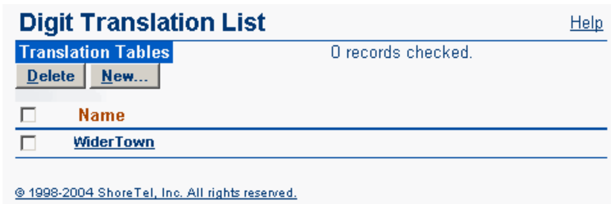


Figure 2-4 Digit Translation List Page

Step 3 Create a new translation profile as follows:

Step a Click the **New** button to add a new entry. The Table Entries page appears as shown in Figure 2-5.

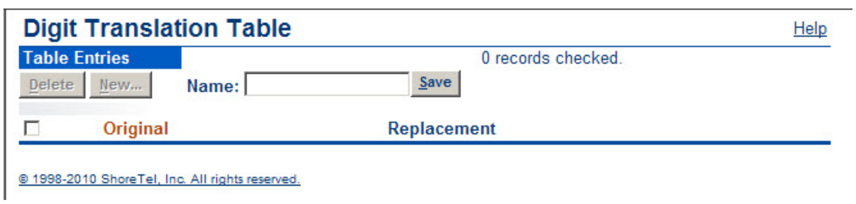


Figure 2-5 Digital Translation Table Entries Page

Step b In the **Name** field, enter the name that you want to use to identify this translation profile and click **Save**. The New button is activated.

Step c Click the **New** button. The Digit Translation Info dialog box appears as shown in Figure 2-6.

Figure 2-6 Digit Translation Page and Entry Dialog Box

Step d In the **Original Digits** field, enter the string that you want translated.

Step e In the **Replacement Digits** field, enter the string that you want to use to replace the original string.

Step f Click **Save**.

Translation table lists appear in profiles for trunk groups and application servers when Simplified Message Desk Interface (SMDI) is selected as the voice mail interface.

2.2.2 Deleting Translation Tables

To delete a digit translation table, do the following:

Step 1 Launch ShoreTel Director.

Step 2 In the ShoreTel Director menu, click **Administration > System Parameters > Digital Translation Tables**. The Digital Translation List page appears as shown in Figure 2-4 on page 40.

Step 3 Check the box next to the digit translation table that you want to delete.

Step 4 Click the **Delete** button.

2.3 Security

2.4 System Extensions

Services such as voicemail, account codes, auto-attendant, Make-me conferences, and ShoreTel conferences associated with the ShoreTel HQ site have system-wide application. These services, when enabled on the HQ site, are automatically assigned an extension on the HQ server. That extension can be used by any user anywhere on the system to access the service though the service may be executed at the server site that is local for the user.

You can view and modify extensions assigned to these system-wide services using the System Extensions page. To access the System Extension page, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Select **Administrator > System Parameters > System Extension**. The Edit System Extensions page appears as shown in Figure 2-7. Table 2-3 describes the elements that appear in System Extensions page.

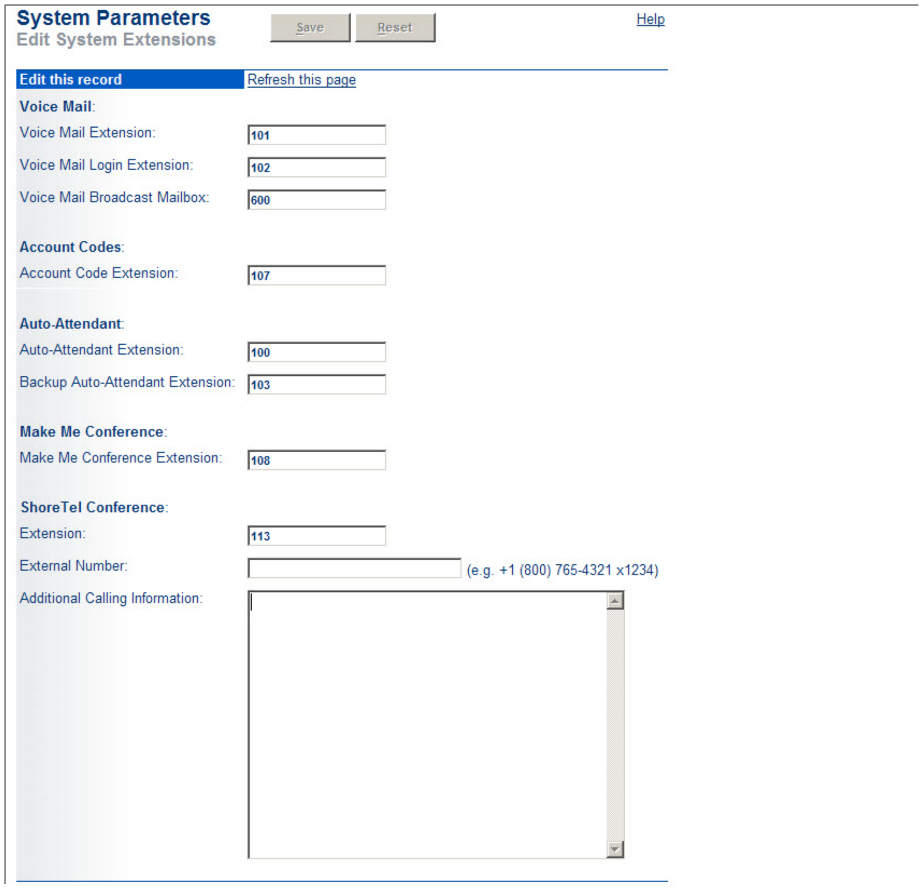


Figure 2-7 System Extensions Edit Page

Table 2-3 System Extension Elements

Element	Description
Voice Mail	This section provides information about system-wide voicemail extensions.
Voice Mail Extension	The extension the system uses for forwarding calls to voicemail.
Voice Mail Login Extension	This is the extension that users use to log into their voice mailbox. It is recommended that users be allowed to dial in from outside the company to retrieve their voicemail. Typically, you direct this number to an auto-attendant menu and configure the menu with a single-digit action of “Transfer to Extension” using the Voice Mail Login Extension parameter.
Voice Mail Broadcast Mailbox	The extension users use to broadcast a voicemail message to all users.
Account Codes	This section provides information about system-wide account code extensions.

Table 2-3 System Extension Elements (Continued)

Element	**Description**
Account Code Extension	This is the extension on the headquarters SoftSwitch associated with the account codes application. When account code collection is optional or forced, calls are routed to this extension for an account code prompt. See also the "Account Codes" on page 237.
Auto-Attendant	This section provides information about system-wide auto-attendant extensions.
Auto-Attendant Extension	The extension used for the system-wide auto-attendant.
Backup Auto-Attendant Extension	The extension you want to use as an auto-attendant backup in case the Headquarters server fails. The backup auto-attendant provides basic inbound call routing in case the auto-attendant on the ShoreTel server is unavailable. In addition, it answers calls routed to voice mail in case voice mail on the ShoreTel server is unavailable. The backup auto-attendant is also used when extensions are not reachable during a network or switch outage and the Admissions Control Bandwidth is exceeded. Callers who are accessing the ShoreTel system over a SIP trunk can access the Backup Auto-Attendant in the same manner as users who are accessing the system via all other trunk types. **ShoreTel** supports RFC2833 (DTMF), so if the voice-mail server is down, external callers can enter an extension using DTMF to ring the extension of the user they are trying to reach.
Make Me Conference	
Make Me Conference Extension	This extension allows users to create conferences on a ShoreTel voice switch. Conferences can have up to six participants depending on how the conference capability is configured.
ShoreTel Conference	This section provides information for conferences conducted using the ShoreTel Service Appliance 100 (SA-100).
Extension	The system-wide extension internal users use to initiate a conference using the SA-100.
External Number	Main external telephone number users can dial to access an SA-100 conference.
Additional Calling Information	Allows you to specify other external telephone numbers users can use to access SA-100s conferences. These numbers may be local to remote sites.

2.5 SNMP

The ShoreTel voice switches support Simple Network Management Protocol (SNMP) agents for the Ethernet interface. These agents provide Management Information Base II (MIB-II) statistics and allow the ShoreTel voice switches to be integrated into standard network management applications.

ShoreTel has tested and supports the HP OpenView network management console.

ShoreTel recommends that you configure your SNMP management station to launch ShoreTel Director automatically when you click a ShoreTel device.

To enable SNMP for your ShoreTel system, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > System Parameters > SNMP**. The Edit SNMP page appears as shown in Figure 2-8.

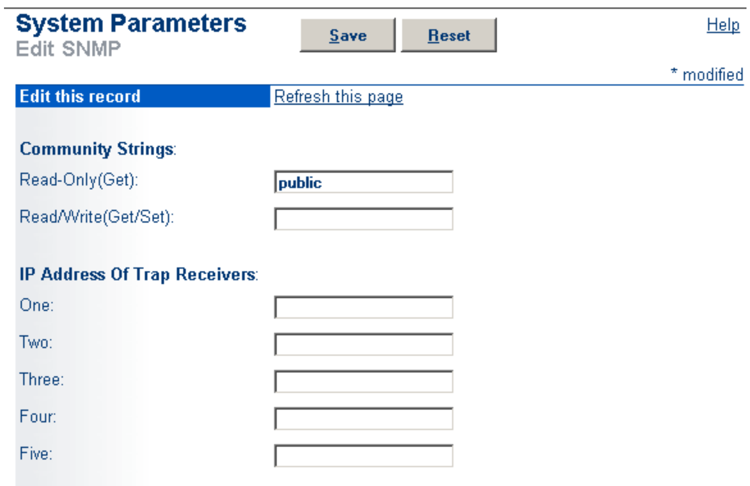


Figure 2-8 SNMP Edit Page

Step 3 In the **Read-Only (Get)** field, enter the SNMP community string your network uses for read-only SNMP messages.

Step 4 In the **Read/Write (Get/Set)**, enter the SNMP community string your network uses for read and write SNMP messages.

Step 5 In the IP Address of Trap Receivers section, type the IP address of each destination that should receive SNMP traps. Up to five locations can receive traps. The destination IP address must have an SNMP trap listener (UDP Port 162) installed.

2.6 BOOTP Server

The BOOTP Server option lets you assign static IP addresses to ShoreTel voice switches. This section describes how you can use the BOOTP option to assign static IP addresses to voice switches.

The requirements for this task are as follows:

- Valid IP address for the location where the switch is to be installed.
- MAC address located on the label on the back panel of the switch.

To use BOOTP Server to assign a voice switch an IP address, do the following:

Step 1 Connect the voice switch to the network.

> **NOTE** For best results when assigning the IP address, we recommend that you connect the switch to the network segment that the server is on.

Step 2 Launch ShoreTel Director.

Step 3 Click **Administration > System Parameters > BOOTP Server**. The Edit SNMP page appears as shown in Figure 2-8. The Edit BOOTP Server page appears as shown in Figure 2-9.

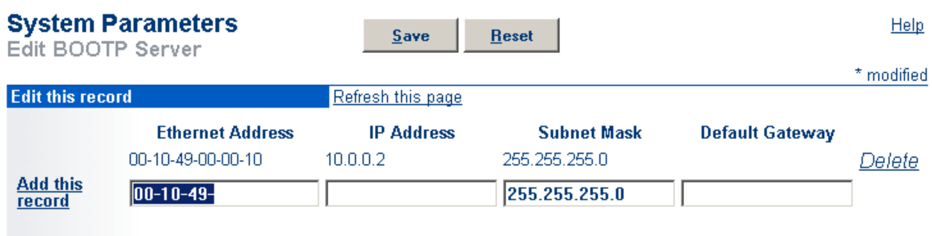


Figure 2-9 BOOTP Server Edit Page

Step 4 In the field under Ethernet Address, enter the MAC address of the switch to which you want to assign the IP address.

Step 5 In the field under IP Address, enter the IP address that you want to assign to the switch.

Step 6 In the field under Subnet Mask, enter the subnet mask that you want to use for the switch.

Step 7 In the field under Default Gateway, enter the IP address of the router or gateway that you want the switch to use to access the network.

Step 8 Click **Save** to save this profile.

The BOOTP server sends the IP address information when the ShoreTel voice switch sends a request.

2.7 Other

A mix of system-wide site parameters are configured using the Other option in the System Parameters menu. System parameters you can configure address passwords, server storage, Instant messaging, ShoreTel upgrades, Active Directory and more.

To access the Other page, do the following.

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > System Parameters > Other**. The Edit Other Parameters page appears as shown in Figure 2-10. Table 2-4 provides information about the elements that appear in the page.

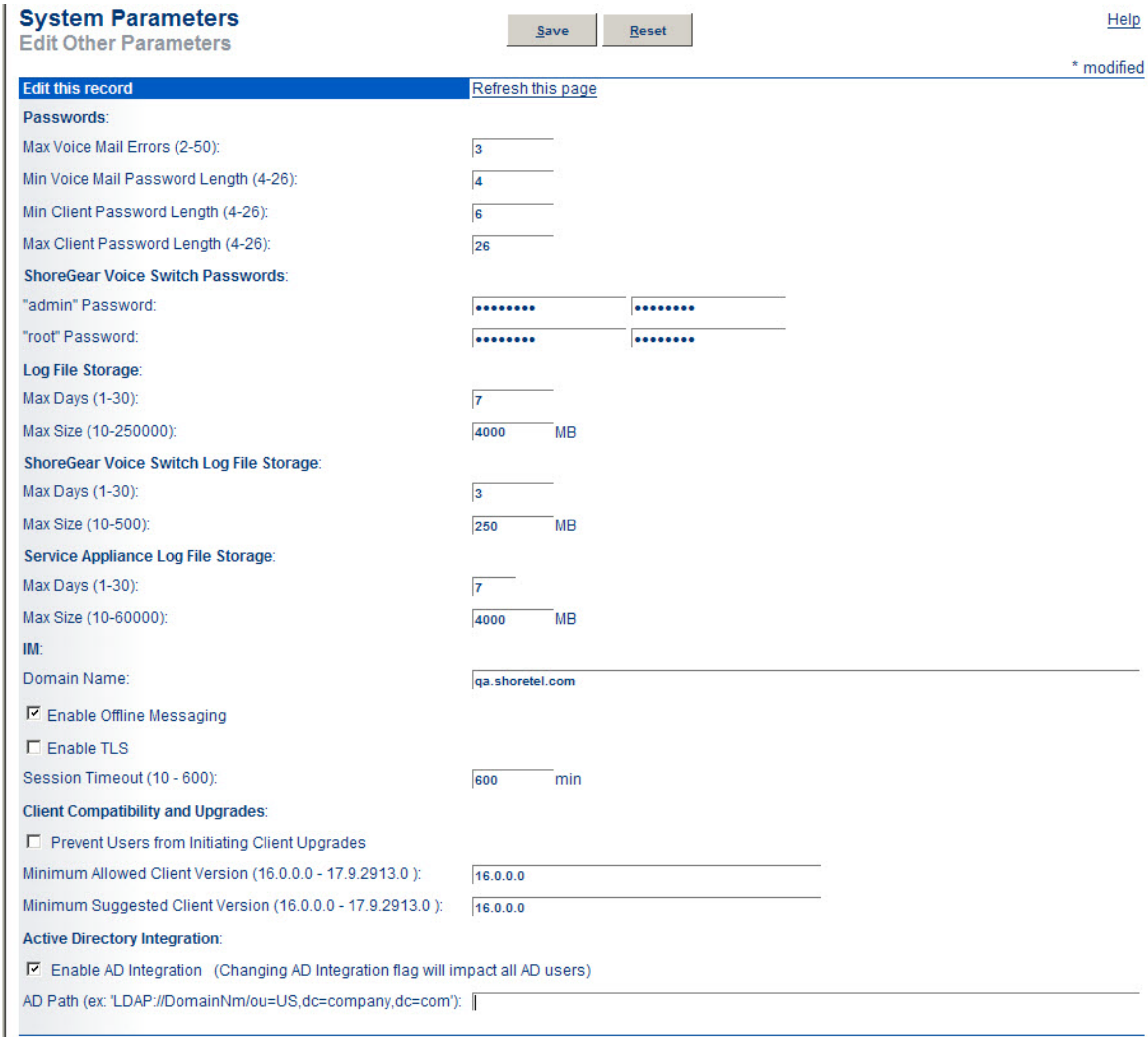


Figure 2-10Other Parameters Edit Page

Table 2-4 Elements in Edit Other Parameters Page

Element	Description
General	
Max Voice Mail Errors (2-50)	Lets you specify how many times a caller can make an error when using the telephone to log into voicemail. When the user exceeds the number of errors, the system informs the user of the error and hangs up.
Min Voice Mail Password Length (4-26)	Lets you specify the minimum number of digits required for voicemail passwords.

Table 2-4 Elements in Edit Other Parameters Page (Continued)

Element	Description
Min Client Password Length (4-26)	These parameters specify the minimum number of characters for the client password needed for a user to log into the ShoreTel Communicator application or ShoreTel Director.
Max Client Password Length (4-26)	Lets you specify the maximum number of characters that users can have in their passwords.
ShoreTel Voice Switch:	
“admin” Password	Lets you change the password use to access the admin account for ShoreTel voice switches.
“root” Password	Lets you change the password used to access the root account for ShoreTel voice switches.
Log File Storage	Lets you set universal parameters ShoreTel servers use for storing log files.
Max Days (1-30)	Lets you specify the number of days servers keep a log file before deleting it.
Max Size (10-500)	Lets you specify the maximum number of megabytes that a server can use to store log files.
ShoreGear Voice Switch Log File Storage	Lets you set universal parameters ShoreTel voice switches use for storing log files.
Max Days (1-30)	Lets you specify the number of days voice switches keep a log file before deleting it.
Max Size (10-600)	Lets you specify the maximum number of megabytes that a switch can use to store log files.
Service Appliance Log File Storage	Lets you set universal parameters service appliances use for storing log files.
Max Days (1-30)	Lets you specify the number of days Service Appliances keep a log file before deleting it.
Max Size (10-60000)	Lets you specify the maximum number of megabytes that a Service Appliance can use to store log files.
IM	Lets you set system-wide parameters that are used for instant messaging.
Domain Name	The domain name (FQDN) that is used for IM.
Enable Offline Messaging	Check to retain messages to users who are off-line. Users will be able to view these messages when they come on-line. When unchecked, messages are dropped when there is no user on-line to receive them.
Enable TLS	Check to allows encryption using transport layer security (TLS).
Session Timeout (10-600)	Lets you specify the duration in minutes that the system allows a message to remain open without a response before timing out.
Client Compatibility and Upgrades	

Table 2-4 Elements in Edit Other Parameters Page (Continued)

Element	Description
Prevent Users from Initiating Client Upgrades	Check the check box to disallow clients to upgrade their Communicator application without receiving an upgrade notification. **NOTE** We recommend setting this parameter when you use the Silent Client Upgrade feature in tandem with Microsoft Active Directory to install client software on remote machines.
Minimum Allowed Client Versions	Enter the earliest version number of Communicator that you will allow users to use. Users are notified when their Communicator software is out of date and are required to upgrade. The default value is the earliest version of Communicator the system software will accept.
Minimum Suggested Clint Version	Enter the earliest version number of Communicator that you want users to use. Users are notified once when their Communicator software is out of date but are not required to update. The default value is the earliest version the installed software will accept.
Active Directory Integration	
Enable AD Integration	Check to allow the system to use Active Directory for authentication.
AD Path	Enter the path that your system uses for the Active Directory.

2.8 Client Compatibility

The Client Compatibility feature provides a way for organizations to have greater control over which versions of Communicator should be deployed during a ShoreTel system upgrade. This feature is designed to lessen the impact of system upgrades for customers. Client Compatibility allows you to upgrade your servers first then your clients. Spreading the upgrade out over time is less demanding on the IT staff and allows users to upgrade at their own convenience.

The Client Compatibility feature lets you specify the earliest version of Communicator that the system supports and suggests an earliest version that clients can use without upgrading. System Administrators can implement the V Minus 1 Compatibility functionality through ShoreTel Director by configuring two system-wide settings.

2.8.1 Implementing Client Compatibility

To implement Client Compatibility, follow these steps:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > System Parameters > Other**. The System Parameters Edit Other Parameters page appears as shown in Figure 2-10 on page 46.

Step 3 Scroll down to the Client Compatibility and Upgrade section and do the following:

- Check the **Prevent Users from Initiating Client Upgrades** checkbox to hide the client upgrade option in Communicator. User are only able to upgrade when a notification is sent.
- In the **Minimum Allowed Client Version** field, enter the version number of the earliest version of Communicator that you want to allow clients to use. The default value is the earliest version your system software will support.

NOTE The earliest and latest versions of Communicator that your ShoreTel system software will support are listed in the parenthesis.

- In the **Minimum Suggested Client Version** field, enter the build number of the earliest version of Communicator that you want to allow clients to use. The user is sent an upgrade message when their version of Communicator falls out of compliance.

NOTE Minimum Suggested Client Version is disabled when the Prevent Users from Initiating Upgrades option is checked.

NOTE The earliest and latest versions of Communicator that your ShoreTel system software will support are listed in the parenthesis.

Step 4 Click the **Save** button to save your changes.

The user is notified once when their version of Communicator falls below the minimum suggested version but is later than the minimum allowed version. A dialog box appears with the upgrade notification allowing them to upgrade immediately. If users chooses to upgrade later, they must use the Upgrade function found in the Communicator Help menu.

2.9 Languages

ShoreTel supports multiple languages and allows you to choose the languages that ShoreTel uses. Languages can be configured for the following:

- Sites
- Auto-Attendant Menus
- Users
- Workgroups
- Route Points
- Trunk Groups

To specify the languages that you want to make available for the system to use, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > System Parameters > Languages**. The Languages page appears as shown in Figure 2-11.

Figure 2-11Languages List Page

Step 3 Check the checkbox in the Enable column for each language that you want to make available on the system.

NOTE You must have a license for each language that use.

2.10 Licenses

This section describes ShoreTel licenses.

2.10.1 Supported Licenses in ShoreTel 12

ShoreTel 12 supports the following ShoreTel licenses:

- ShoreWare System License (Enterprise and Small Business Editions)
- ShoreWare Additional Site License
- ShoreWare Extension License
- ShoreWare Mailbox License
- ShoreWare SoftPhone License
- ShoreWare Additional Language License
- Mobile Access License
- ShoreWare SIP Trunk License
- ShoreWare SIP Phone License
- ShoreWare Standard Resolution Video License
- ShoreWare High Resolution Video License
- Professional Access License
- Operator Access License
- Workgroup Agent Access License
- Workgroup Supervisor Access License
- ShoreWare External Unified Messaging SIP Link License
- Audio Conference License
- Web Conference License

Self-audited Licenses:

- Personal Access License

- ShoreWare Remote Server Software
- ShoreWare TAPI Application Server
- ShoreWare Phone API License

2.10.2 Viewing Your ShoreTel Licenses

To view the installed ShoreTel licenses in ShoreTel Director:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > System Parameters > Licenses > Requirements.** The License Requirements page shown in Figure 2-12 appears.

License Requirements

License Requirement List — Register and Request System Key

Name	Configured	Purchased
Keyed Licenses:		
ShoreWare System License (Enterprise Edition)	1	1
ShoreWare Additional Site License	2	4
ShoreWare Extension License	13	100
ShoreWare Mailbox License	13	100
ShoreWare SoftPhone License	5	5
ShoreWare Additional Language License	0	5
ShoreWare Remote Web Reporting License	1	1
Mobile Access License	4	5
ShoreWare SIP Phone License	0	0
ShoreWare SIP Trunk License	0	0
ShoreWare Standard Resolution Video License	0	0
ShoreWare High Resolution Video License	0	0
Operator Access License	0	0
Professional Access License	11	50
Workgroup Agent Access License	0	50
Workgroup Supervisor Access License	0	50
ShoreWare External Unified Messaging SIP Link	0	0
Audio Conference License	50	50
Web Conference License	50	50
Self-audited Licenses:		
Personal Access License	2	20
ShoreWare Remote Server Software	1	2
ShoreWare TAPI Application Server	1	2
Shoreware Phone API License	2	5

Figure 2-12License Requirements Page

The License Requirements page lists licenses that are available for the ShoreTel system and the quantity of licenses you have acquired. You can use this page to track and manage all licenses.

The licenses are divided into keyed and self-audited licenses. Self-audited licenses do not have a key associated with them. They are tracked on the license page as a tool to assist system administrators in tracking the number required based on the current configuration versus the number that have been purchased, which they enter manually.

Five Communicator licenses, each of which corresponds to a Communicator type, span the Communicator feature set. ShoreTel defines four of these licenses as Keyed Licenses.

2.10.3 Compliance

If your system is out of compliance, ShoreTel Director offers 45-days to comply with the license requirements by either removing unneeded configurations and/or by ordering additional licenses. The 45-day grace period allows you to make ad hoc, unplanned changes that could temporarily exceed your license limits, but gives you time to get back into compliance.

WARNING Do not upgrade unless you are already in license compliance. If you upgrade and you are out of compliance, you will only have 45 days before being locked out of ShoreTel Director. Contact your ShoreTel Partner or ShoreTel Installed Base Business Services Team at Shorecare_admin@shoretel.com if you have any outstanding license issues.

2.10.4 Registering the ShoreTel Software

You must register your ShoreTel system software and obtain a system key before you can use the system. This is done using ShoreTel Director. After you registered and apply for licenses, ShoreTel acknowledges your submission in an email and mails your system key within 3-5 days. You have 45 days to install your system key.

IMPORTANT After upgrading to a new version, you will have 45 days to ensure the new key is installed. If it is not installed, you will be locked out of ShoreTel Director.

To request a register the software and request a system key:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > System Parameters > Licenses > Requirements.** The License Requirements page shown in Figure 2-12 on page 51 appears.

Step 3 Click the **Request System License Key** button. The Edit Contact page appears as shown in Figure 2-13.

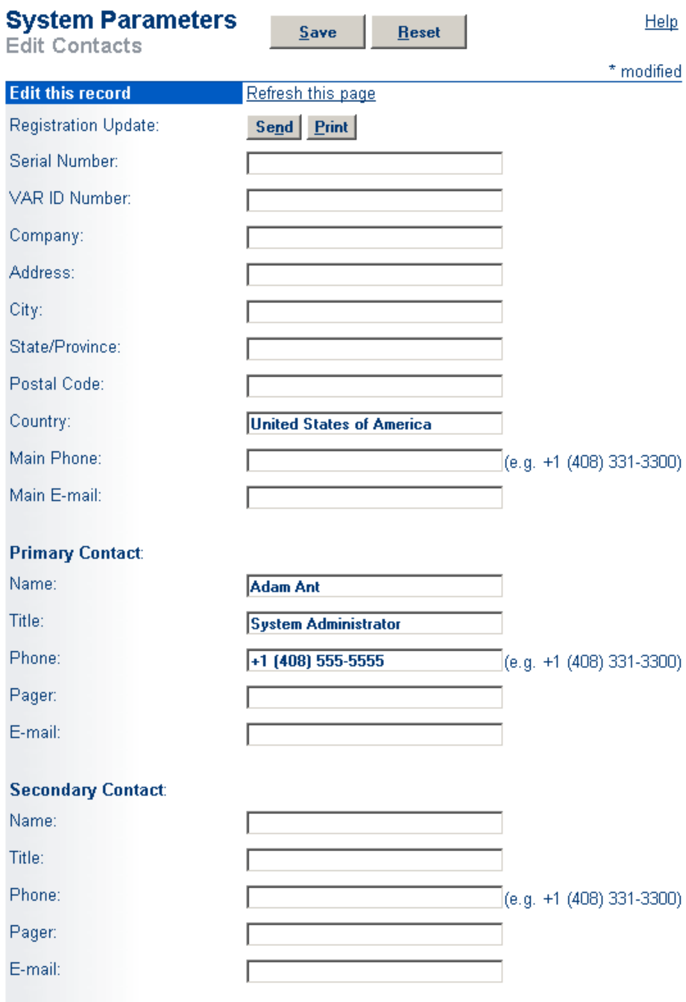


Figure 2-13System Parameters Edit Contacts Page

Step 4 As a minimum, enter the information requested in the “Register and request system key” and Primary Contact sections. Be sure to include the sales order number for your purchase. Figure 2-5 describes the elements that appear on the Edit Contact page.

Step 5 Click the **Now** button at the top of the form to send in your request.

Table 2-5 Elements on the Edit Contact Page

Element	Description
Partner Name	Name of the partner or reseller from whom the system was purchased. This field is required.
Customer Name	Name of the customer. This field is required.
Address	Address of the customer. This field is required.
City	City where the customer is located. This field is required.
State/Province	State or province where the customer is located. This field is required.

Table 2-5 Elements on the Edit Contact Page (Continued)

Element	Description
Postal Code	Postal code for the customer. This field is required.
Country	Country where the customer is located. This field is required.
Main Phone #	Main telephone number of the customer.
Main E-mail	Main e-mail address of the customer.
Primary Contact	
Name	Name of the administrator responsible for the ShoreTel system. This field is required.
Title	Job title of the administrator responsible for the ShoreTel system. This field is required.
Phone	Phone number of the administrator responsible for the ShoreTel system. This field is required.
Pager	Pager number of the administrator responsible for the ShoreTel system. This field is required.
E-mail	E-mail address of the administrator responsible for the ShoreTel system. This field is required.
Secondary Contact	
Name	Name of the back up administrator ShoreTel can contact regarding the ShoreTel system.
Title	Job title of the aback up administrator ShoreTel can contact regarding the ShoreTel system.
Phone	Phone number of the back up administrator ShoreTel can contact regarding the ShoreTel system.
Pager	Pager number of the back up administrator ShoreTel can contact regarding the ShoreTel system.
E-mail	E-mail address of the back up administrator ShoreTel can contact regarding the ShoreTel system.

2.10.5 Installing License Keys

To install license keys, do the following:

Step 1 View the license packet that you received from ShoreTel.

Step 2 Launch ShoreTel Director and in the ShoreTel Director menu click **Administration > System Parameters >Licenses > Keys**. The License Key Info dialog box appears.

Step 3 Click the **New** button at the top of the page. The License Key Info dialog box appears as shown in Figure 2-14.

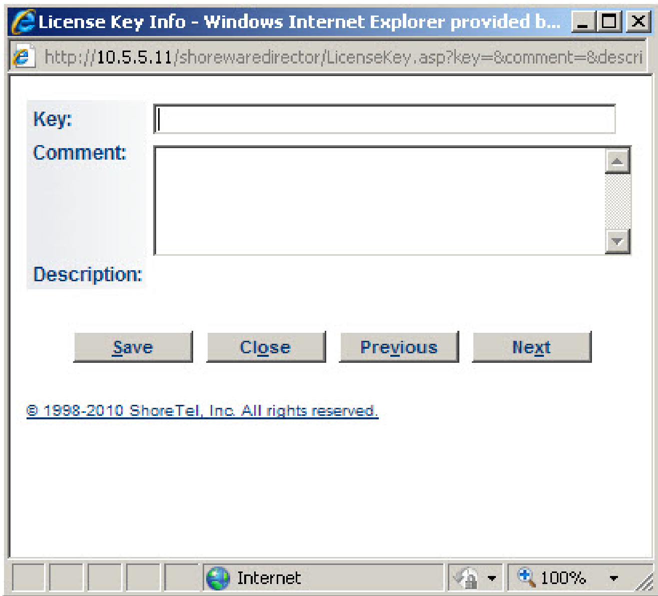


Figure 2-14License Key Info Dialog Box

Step 4 In the Key field, enter the license key that you received from ShoreTel.

Step 5 In the Comment field, enter a description of the license.

Step 6 Click **Save**. The license activates and the information appears in the License Key page.

2.10.6 Keyed License Types

Keyed licenses are added by entering a license key string obtained from the reseller or vendor. Embedded in the license key are the type and number of licenses associated with that key. Once a valid key is entered, the system decodes it and details the type and number of licenses added. Keyed licenses are additive and more than one can be entered into ShoreTel Director over time.

These licenses are counted on the License Requirements page in Director, the Keyed Licenses section. The licenses are grouped according to the following categories:

- **ShoreWare System License:** This count includes licenses required on a per-system basis.
- **ShoreWare Additional Site License:** This count includes licenses required for each site beyond the main headquarter location. For installed base customers, when you upgrade and request your new system key, you will automatically receive additional site licenses for all configured sites.
- **ShoreWare Extension License:** This count includes all extensions licensed by both Extension Only and Extension and Mailbox licenses.
- **ShoreWare Mailbox License:** This count includes all mailboxes licensed by both Mailbox Only and Extension and Mailbox licenses.
- **ShoreWare SoftPhone License:** This count includes SoftPhone licenses, which are issued on a per-user basis. Obtain and install one licence for each SoftPhone user.

- **ShoreWare Additional Language Licenses:** This count includes licenses if more than one language is enabled.
- **ShoreWare Communicator for Mobile License:** This count includes licenses required for each client that is enabled for Communicator for Mobile.
- **ShoreWare Personal Communicator License is the default client configuration. It does not require a special license. This provides the following access level to all users:**
  - — Desktop call control.
  - — Visual voice mail.
  - — Call history.
  - — Directory services.
  - — Options to control call handling and message notification.
- **ShoreWare Professional Communicator License** is a keyed license provides access to the following:
  - — All functions available through Personal Communicator.
  - — Instant Messaging Presence
  - — Contact Viewer
- **ShoreWare Workgroup Agent Communicator License** is keyed license that provides access to the following:
  - — All functions available through Professional Communicator.
  - — Ability to transfer calls by dragging and dropping call cells into the buddy list.
  - — Workgroup access utilities, including login, log out, and wrap up.
  - — Workgroup Queue Monitor.
- **ShoreWare Workgroup Supervisor Communicator License** is a keyed license provides access to the following:
  - — All functions available through Workgroup Agent Communicator.
  - — Workgroup Agent Monitor.
- **ShoreWare Operator Communicator License** is a keyed license that provides access to the following:
  - — All functions available through Workgroup Supervisor Communicator.
  - — Advanced Buddy List functions, including pickup, unpark, edit buddies' active call properties, and edit call notes of buddies.
  - — Bridge Call Appearance Monitor.
- **ShoreWare SIP Phone License** is a keyed license that enables the system to support one SIP device through a SIP proxy.
- **ShoreWare Standard Resolution Video License** is a keyed license that enables Communicator to support one point to point video session at VGA resolution (640x480).
- **ShoreWare High Resolution Video License** is a keyed license that enables Communicator to support one point to point video session at XGA resolution (1024x768).

2.10.7 Extension and Mailbox Licenses

Systems require one extension license for each configured extension-only user. If more than one key is installed, the number purchased is the sum of all valid keys. Extension-only users have an extension but no ShoreTel mailbox. They may have external mailboxes that can accessed using Simplified Message Desk Interface (SMDI).

Systems require one Mailbox license for each configured Mailbox-only user. If more than one key is installed, the number purchased is the sum of valid keys. Mailbox-only users are only those users with ShoreTel mailboxes that may use SMDI.

Systems require one Combo license for each user configured for Extensions and Mailboxes. If more than one key is installed, the number purchased is the sum of all valid keys.

Table 2-6 lists the features are available through Extension, Mailbox, and Combo licenses.

Table 2-6 Licensed Extension and Mailbox Feature Availability

Feature	Combo	Extension Only	Mailbox Only
		Includes 3rd-party SMDI-based VM to ST PBX	**Includes ST SMDI-Based VM to 3rd-party PBX**
PBX features			
Use SoftPhone (requires SoftPhone license)	Yes	Yes	No
Make call, take call, etc.	Yes	Yes	No
Voicemail features			
Configure call handling modes	Yes	Yes	Yes
Forward calls to configured destination	Yes	Yes	No[a]
Create and play greetings	Yes	No	Yes
Use the Personal Assistant	Yes	No	Yes
Notification escalation	Yes	No	Yes
Configure Find Me	Yes	No	Yes
System call handling schedule	Yes	No	Yes
Create call handling notes	Yes	Yes	Yes
Assign Extension	Yes	No	Yes
Record name	Yes	No	Yes
Automated attendant features			
Dial by number, name	Yes	Yes	Yes[b]
Transfer to / Go to extension	Yes	Yes	YesTable b
Message by number, name	Yes	Yes	Yes
Advanced features			
Extension Assignment	Yes	Yes	No
Member of a hunt group	Yes	Yes	No

Table 2-6 Licensed Extension and Mailbox Feature Availability (Continued)

Feature	Combo	Extension Only Includes 3rd-party SMDI-based VM to ST PBX	Mailbox Only Includes ST SMDI-Based VM to 3rd-party PBX
Member of a workgroup	Yes	Yes	No
Communicator features			
Communicator: Standard, Professional, Workgroup Agent, Workgroup Supervisor, Operator	Yes	No Mailbox Features	No extension features
Extension monitor	Yes	Operator only features	No
Agent monitor	Yes	No mailbox features	No
Queue monitor	Yes	No mailbox features	No
voicemail viewer	Yes	No	Yes
Call history	Yes	Yes	No
System directory	Yes	No mailbox features	No extension features
Outlook features			
Fwd voicemail as wav attachment	Yes	No	Yes
voicemail form integration	Yes	No	Yes
Outlook Contact/Quick Dialer	Yes	Yes	No
Outlook Contact/Screen Pop	Yes	Yes	No
Outlook Calendar integration	Yes	Yes	Yes

a. Although call forwarding is handled by the 3rd party PBX, calls arriving at the ShoreTel voicemail system are routed as specified by ShoreTel voicemail forwarding conditions.
b. Calls will be directed to mailbox only.

2.10.8 Self-Audited Licenses

The licenses listed below are self-audited. If the usage exceeds the current licences, you will be notified until licensed capacities meet or exceed usage.

- **ShoreTel Personal Communicator Software:** This count includes the number of ShoreWare Personal Communicator licenses needed.
- **ShoreWare Remote Server Software:** This count includes licenses that correspond to additional ShoreWare servers, defined in Director, that correspond to additional voicemail servers. Up to 20 additional ShoreWare remote servers can be configured beyond the initial or headquarters server.
- **ShoreWare SIP Trunk:** This count includes licenses necessary for the implementation of SIP trunks.
- **TAPI Application Server:** This count includes licenses for remote TAPI Application Servers that have the “Allow Voice Mailboxes” check box deselected. The number purchased should match the number of deprioritized servers that exist at a particular site.

- **ShoreWare Phone API License:** This count includes licenses for the Phone API. (For more information, contact ShoreTel Professional Services for the appropriate SDK document.)

The license status page has been enhanced to easily be printed or sent via e-mail for purposes of license compliance verification. No license status will be transmitted without explicit action on the part of the administrator.

2.10.9 Sending Contact Information

After providing the required information on the System Parameters Edit Contacts page, you can send it to ShoreTel, Inc. in one of two ways—by email or by regular mail.

- To send the registration information by email, click Send. This generates an e-mail message to registration@shoretel.com. It also requires that the SMTP service on the server be properly configured and that the server be connected for email. You can resend the contact information at any time by updating the page and clicking Send.

  The MAC address for each ShoreTel voice switch is also included in the registration e-mail.

- To print the registration information for mailing via regular mail, click Print. Mail the registration information to the following address:

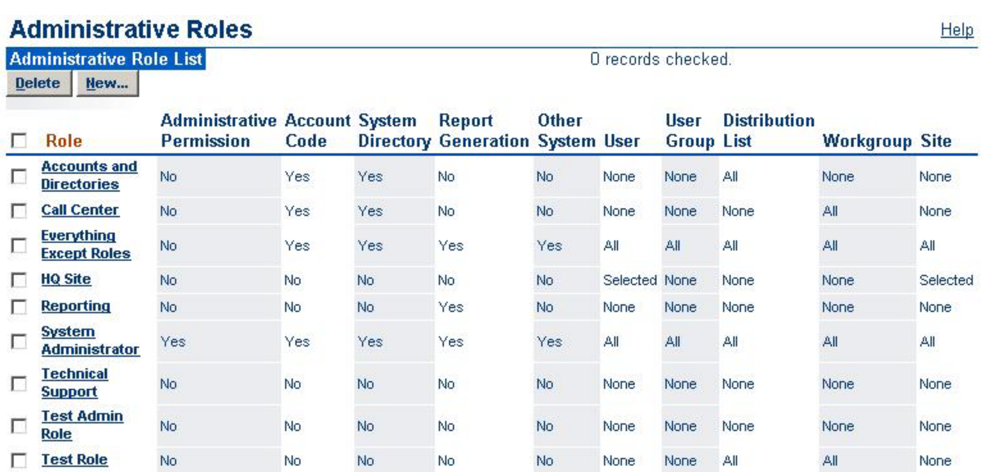
  Global Support Services — Product Registration
  ShoreTel, Inc.
  960 Stewart Drive
  Sunnyvale, CA 94085

2.11 Administrative Permissions

The Administrator Permissions pages allow the System Administrator to assign and delegate administrative roles to users at one or more sites. Expand the Administrative Permissions link to see all the administrative links. They include:

- Roles
- Administrators

Click the Roles link to see the Administrative Roles list page. This page shows each Administrative Roles that have been created and summarize their permissions.



Role	Administrative Permission	Account Code	System Directory	Report Generation	Other System	User	User Group	Distribution List	Workgroup	Site
Accounts and Directories	No	Yes	Yes	No	No	None	None	All	None	None
Call Center	No	Yes	Yes	No	No	None	None	None	All	None
Everything Except Roles	No	Yes	Yes	Yes	Yes	All	All	All	All	All
HQ Site	No	No	No	No	No	Selected	None	None	None	Selected
Reporting	No	No	No	Yes	No	None	None	None	None	None
System Administrator	Yes	Yes	Yes	Yes	Yes	All	All	All	All	All
Technical Support	No	No	No	No	No	None	None	None	None	None
Test Admin Role	No	No	No	No	No	None	None	None	None	None
Test Role	No	No	No	No	No	None	None	All	All	None

Figure 2-15The Administrative Roles List Page

Clicking the New button or an administrative title from the Role column invokes the Edit Administrative Roles page (see Figure 2-16). From here, you can define a new administrative role or change the permissions for an existing role.

To delete a role, select the check box to the left of the entry and then click ***Delete***. Note that if the last role with Administrative Permissions Management enabled is removed, then the default **admin** account (as created during initial installation) is re-activated and given complete administrative permissions.

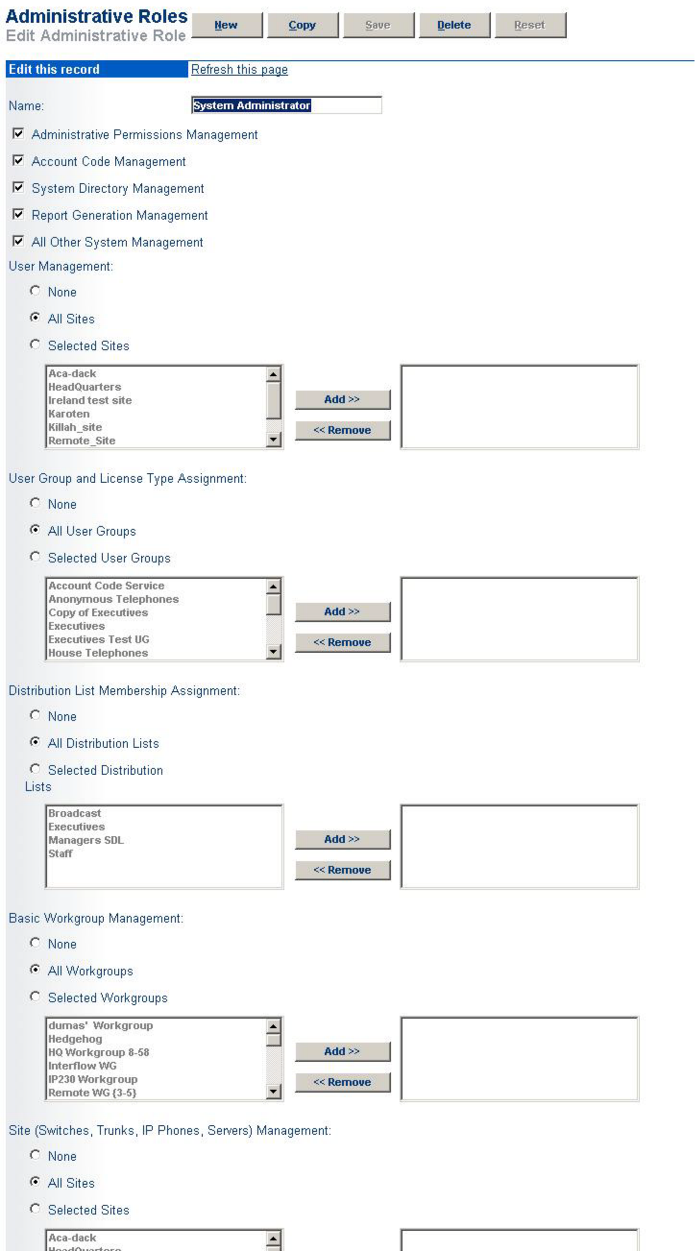


Figure 2-16Adding or Editing Administrative Roles

2.11.1 Parameters

Click parameters to enable permissions. Permissions are additive; that is, the more selections, the greater the permissions. Select as many or as few as are needed for the administrative role being defined. For example, a company with one system administrator may have all parameters turned on. As another example, an administrative assistant may have permission to change Distribution Lists at one site.

- **Name:** This is the name of the Administrative Role.
- **Administrative Permissions Management:** This check box assigns permission to create new administrative roles and to assign them to any and all levels of user. This is a powerful permission and should be limited to your lead administrator(s).
- **Account Code Management:** This check box assigns permission to add, change, and delete Account Codes for all sites. As an example of specialized use, very often a department other than Information Technology wants to manage account codes and needs no other permissions. This permission is granted for all sites.
- **System Directory Management:** This check box assigns permission to add, change, and delete entries in the System Directory. This permission is granted for all sites.
- **Report Generation Management:** This check box assigns permission to generate Call Detail Record (CDR) reports via Director from a local host or a remote server.
- **All Other System Management:** This check box controls permission to set dialing plans, system-wide extensions, including route point and workgroup extensions, sites, IP phone options, digit translation tables, voicemail options, auto-attendant options and schedules, user groups, trunk groups, local prefixes, DNIS digit maps, BOOTP server, classes of service, call control, system parameters such as password length, AMIS options, call handling defaults, event filters, licenses, extension lists, hunt groups, paging groups, and contact information. This permission is granted for all sites.
- **User Management:** Permission to add, change, and delete users may be granted for all sites or for a set of selected sites. Click All Sites to grant permission system-wide. Click Selected Sites to limit permissions, then highlight the sites to be permitted, and click Add to move them to the permitted list.

  Users whose home ports are at the site(s) selected can be managed by an authorized administrator. This permission allows changes only to users who have no administrative role (that is, for whom none of the four administrative check boxes is checked). Also, changes cannot be made to a user's administrative role. Only Administrative Permissions Management grants permission to change administrative roles.

  Deny permission by clicking None.
- **User Group Assignment:** Permission to add users to or move users between user groups may be granted for all sites or for a set of selected sites. Click All User Groups to grant permission system-wide. Click Selected User Groups to limit permissions, then highlight the user groups to be permitted, and click Add to move them to the permitted list. Be sure to select all groups you may be moving users to or from.

  Permission is not extended to adding, changing, or deleting User Group options and Class of Service settings (an administrator would need All Other System Management permission).

  Note that checking the All User Groups includes all user groups currently existing as well as those created after permission is first granted.

Deny permission by checking None.

- **Distribution List Membership Assignment:** Permission to add or remove users on existing Distribution Lists may be granted for all lists or for a set of selected lists. Click ***All Distribution Lists*** to grant permission system-wide. Click ***Selected Distribution Lists*** to limit permissions, then highlight the lists to be permitted, and click ***Add*** to move them to the permitted list.

  Note that permission to create or delete lists is not granted here (an administrator would need All Other System Management permission).

  Deny permission by checking None.

- **Basic Workgroup Management:** Permission to add or change options for workgroups may be granted for all workgroups or for a set of selected workgroups. Click All Workgroups to grant permission system-wide. Click ***Selected Workgroups*** to limit permissions, then highlight the workgroups to be permitted, and click ***Add*** to move them to the permitted list.

  Workgroup attributes not given change permission with this option include workgroup Name, Extension, Backup Extension, DID, DNIS, User Group, Mailbox, Accept Broadcast Messages, Include in Dial By Name, and Make Number Private (an administrator would need All Other System Management permission).

  Note that checking the All Workgroups includes all workgroups currently existing as well as those created after permission is first granted.

  Deny permission by checking None.

- **Site Management:** Permission to add and alter sites and their related switches, trunks, IP phones, and servers may be granted for all sites or for a set of selected sites. Permission includes access to Quick Look at permitted sites. Permission includes adding and deleting anonymous phones at permitted sites.

  Attributes excluded from permission include Trunk Groups (an administrator would need All Other System Management permission).

  Deny permission by checking None. Click All Sites to enable changes to all sites in the system. Click Selected Sites and Add sites from the list to enable access to less than all sites in the system.

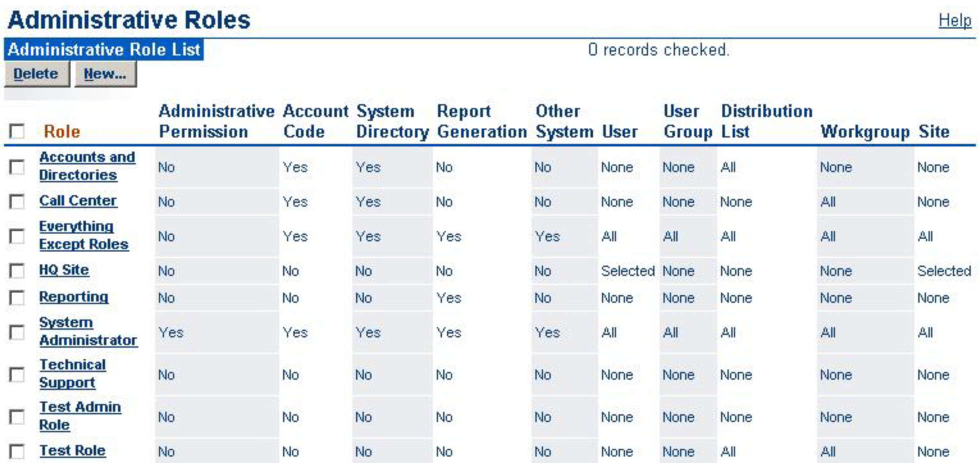
The initial administrator set up during installation has full permissions. When upgrading the ShoreTel system, current System Administrators are granted full permissions. Current Technical Support users have no permission to change parameters but are allowed to read all pages.

For some Director pages where read-only permission is given to some parameters because all parameters on the page may not be changed, the read-only fields will be grayed out.

ShoreTel Director is delivered with the following default Administrative Roles:

- Accounts and Directories
- Call Center
- Everything Except
- HQ Site
- Reporting
- System Administrator
- Technical Support
- Test Admin Role
- Test Role

The various default roles, along with their permissions, are shown below.

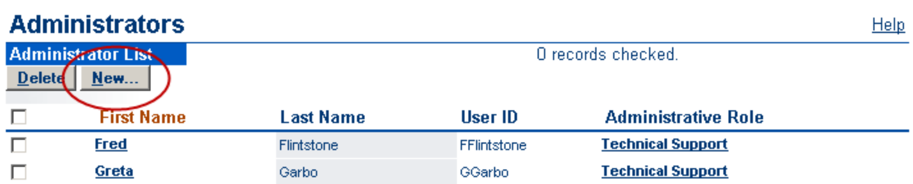


Role	Administrative Permission	Account Code	System Directory	Report Generation	Other System	User	User Group	Distribution List	Workgroup	Site
Accounts and Directories	No	Yes	Yes	No	No	None	None	All	None	None
Call Center	No	Yes	Yes	No	No	None	None	None	All	None
Everything Except Roles	No	Yes	Yes	Yes	Yes	All	All	All	All	All
HQ Site	No	No	No	No	No	Selected	None	None	None	Selected
Reporting	No	No	No	Yes	No	None	None	None	None	None
System Administrator	Yes	Yes	Yes	Yes	Yes	All	All	All	All	All
Technical Support	No	No	No	No	No	None	None	None	None	None
Test Admin Role	No	No	No	No	No	None	None	None	None	None
Test Role	No	No	No	No	No	None	None	All	All	None

Figure 2-17The Administrative Roles List Page

2.11.2 Assigning an Administrative Role to a User

From Administrative Permissions, click the Administrators link to reach the Administrator List page.



First Name	Last Name	User ID	Administrative Role
Fred	Flintstone	FFlintstone	Technical Support
Greta	Garbo	GGarbo	Technical Support

Figure 2-18The Administrator List Page

The Administrator List page (Figure 2-19) shows the administrative role assigned to each user. A user may have only one administrative role assigned. New users are created with no administrative role assigned to them.

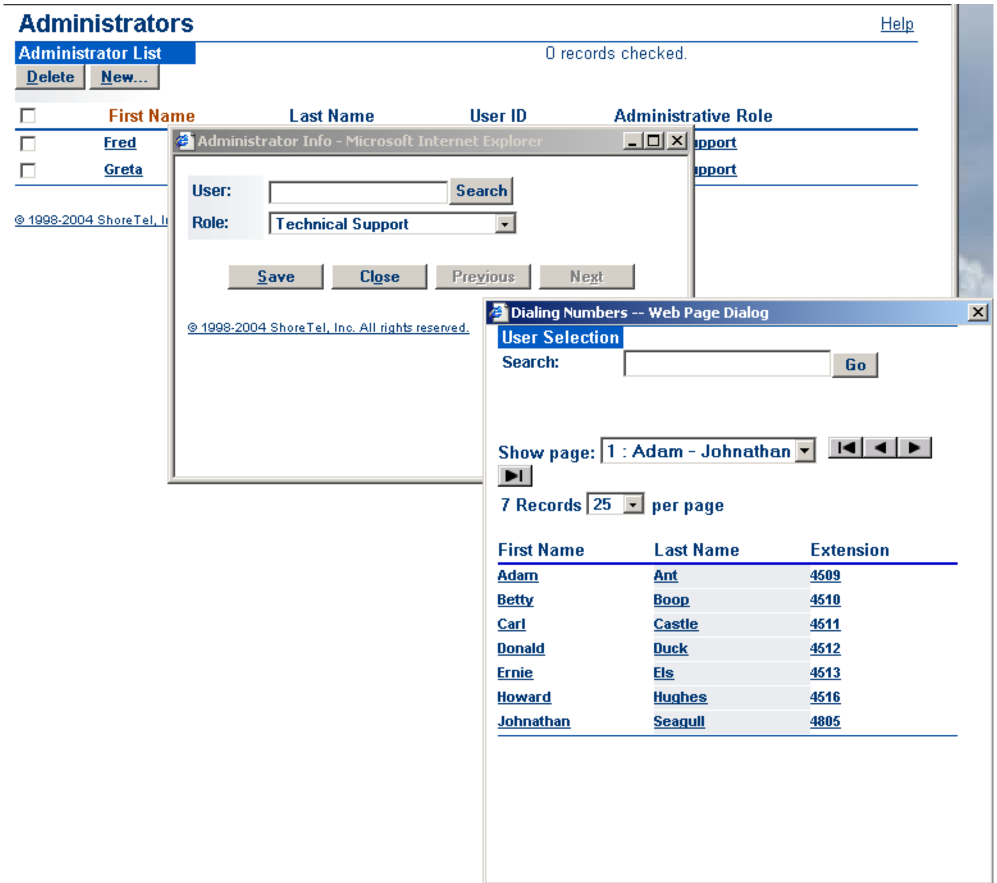


Figure 2-19Assigning an Administrative Role to a User

After defining the various Administrative Roles, you select which users will be assigned which roles.

To assign an administrative role to a user, click New from the Administrators List page. The pop-up lets you type a User name or click Search to select from a list of users. Then assign a Role from the drop-down list.

Users with no administrative role may not log in to Director.

If desired, you can assign users to the “Reporting” administrative role, which will allow them to run web-based CDR reports while preventing them from doing anything else to modify the configurations in ShoreTel Director.

Click Delete on the Administrator List page to delete users from the list.

You may delete a user by checking the box to the left of a name and clicking the Delete button. Note that you cannot delete all users. At least one user must remain on the list to preclude the occurrence of no one being left to administer the system.

CHAPTER 3

ShoreTel Sites

This chapter explains how sites are implemented and configured in a ShoreTel system. The topics discussed include:

- "Overview" on page 65
- "Creating a Site" on page 65
- "Parameters" on page 67

3.1 Overview

The ShoreTel site is an architectural concept designed to help you effectively and efficiently organize your internal telephone environment. Sites allow you to evaluate how your system will be used and then to deploy and setup your equipment accordingly. Sites can be created to accommodate geographical requirements where the external environment affects outbound calls or logical requirements such as a need to separate users given advanced functions from standard users. After a site is created, you can assign to it servers, switches, appliances, users, other sites and so on. Configuring sites lets you assign features to the associated site. For example, a site has a country, a local area code, and a site operator, as well as an admission control setting.

ShoreTel, by default, is configured with one site called "Headquarters."

3.2 Creating a Site

This section describes how to create a site. To create a site, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Site**. The Sites page appears as shown in Figure 3-1.

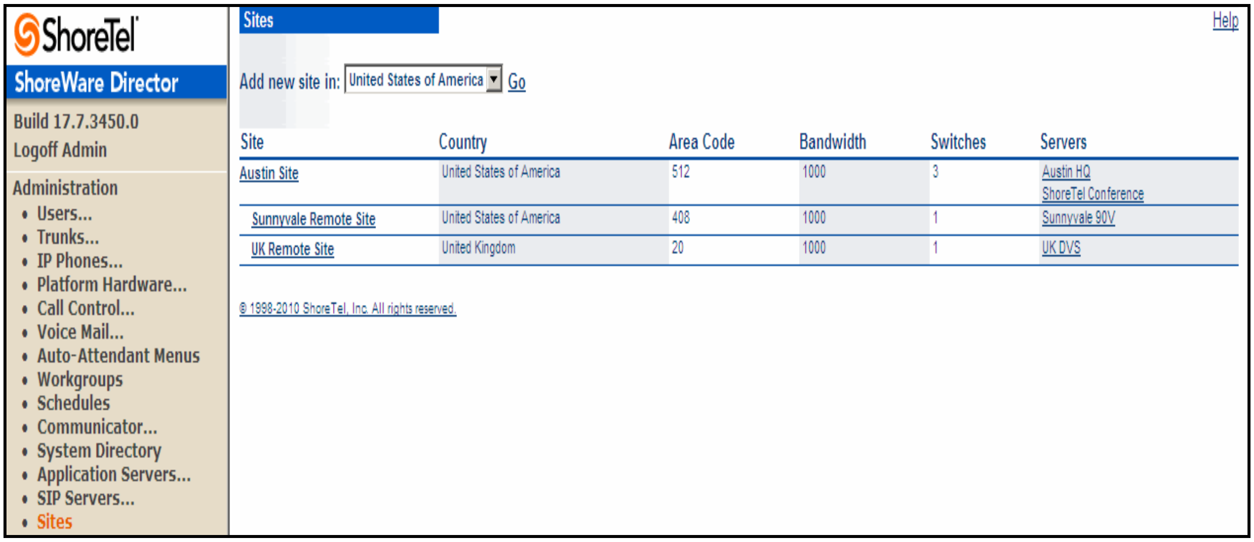


Figure 3-1 Sites Page

Step 3 In the “Add new site in” field, select the country where you want to create the site.

Step 4 Click **Go**. The Edit Site page appears as shown in Figure 3-2.

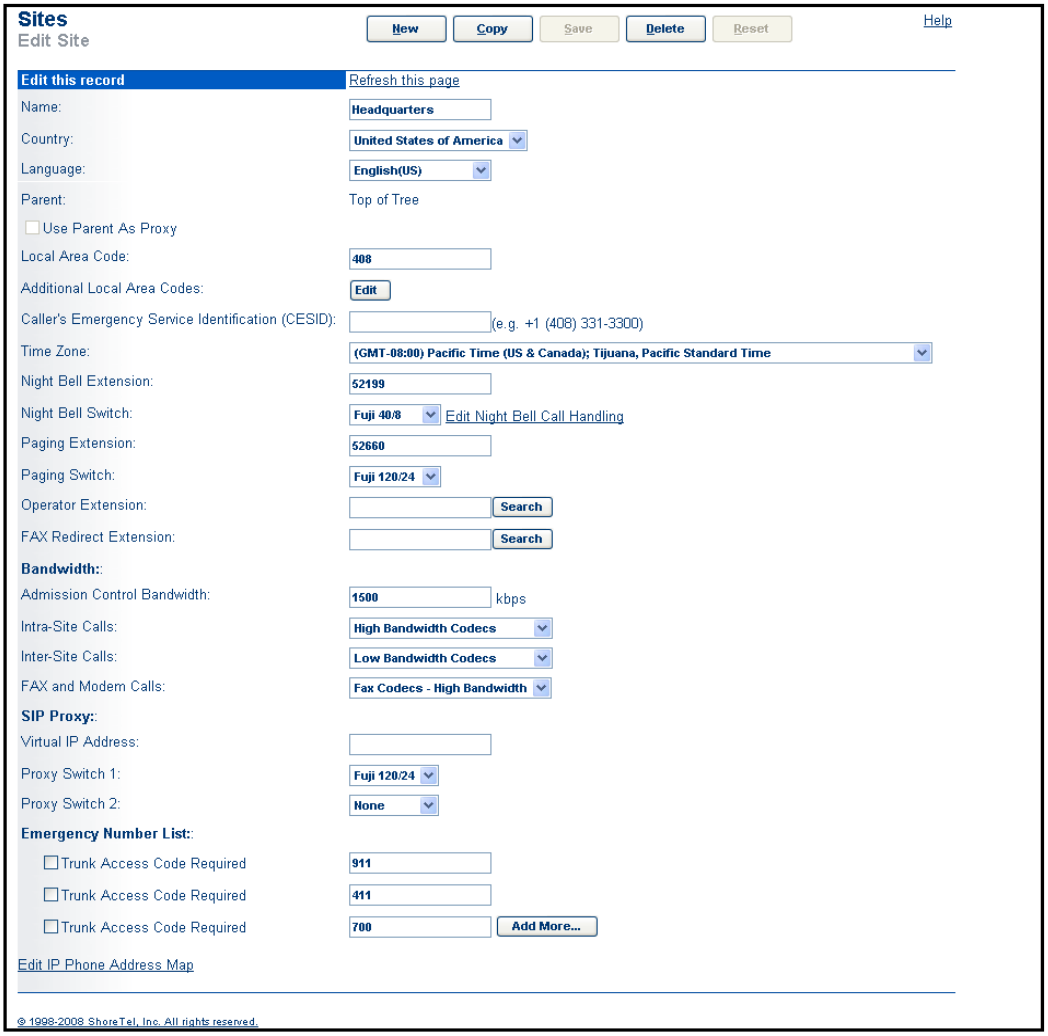


Figure 3-2 Site Edit Page

3.3 Parameters

The parameters on the Sites edit page are as follows:

- **Name:** This is the name of a new or existing site. It must be unique.
- **Country:** This is the name of the country in which the site is located.
- **Language:** This is the default language for the site. You must obtain a license to enable more than one language. For more information see the "Licenses" on page 50.
- **Parent**: The default parent site is Headquarters. Headquarters does not display the drop-down list of sites. Sites other than Headquarters must select a parent. This server is used for two purposes:
  - **By ShoreTel Director** to provide a default server when new users are added
  - **By the call control software** in the ShoreTel voice switches so that it knows where to route calls that request voice mail service

  Only valid parent sites appear in the drop-down list. Child sites and the site currently being edited do not appear.

- **Use Parent As Proxy:** This allows the child site to use the parent site trunk for non-routable calls (911, 611, 011, etc.) if no trunks are available at the child site. The proxy site must be in the same country as the child site.
- **Extension Prefix:** The On-Net Dialing feature enables the division of phone numbers into two separately managed parts, an extension prefix (and similar in concept to a site code) and a user extension. This division offers greater flexibility and facilitates integration with legacy phone systems.

  The Extension Prefix field will not appear in this window until after you have modified the Dialing Plan window, thus enabling the On Net Dialing feature.
- **Local Area Code:** This defines the local area code of the site so that users can dial local numbers without an area code. In the United States, this is the area code used for 7-digit dialing. For example, when the user dials an access code followed by seven digits at the site, this is the area code they are dialing.

  This also defines the area code that is considered "local" from a call permissions point of view.
- **Additional Local Area Codes:** In the United States, this defines area codes that can be dialed using 10-digit dialing instead of 1+10-digit dialing. For example, if the site is in an overlay area with multiple local area codes that require 10-digit dialing, you can be consistent with the dialing plan in your region by entering the additional area codes in this parameter.

  This also defines additional area codes that are considered "local" from a call permissions point of view.
- **Caller's Emergency Service Identification (CESID):** The Caller's Emergency Service ID (CESID) is the telephone number sent to the service provider when a user dials an emergency services number (e.g., 911 in the U.S.). This feature is only applicable to T1 PRI trunks. See Appendix A for more information.
- **Time Zone:** This is the site's time zone that is associated with the ShoreTel switches. It is used to deliver the correct time and date to caller ID telephones.
- **Night Bell Extension:** This is the extension that is used to ring the site's night bell. This extension must be associated with a ShoreTel switch audio output port that you specify as the next parameter. This extension is unique.

  You must configure the appropriate switch before assigning the night bell extension.
- **Night Bell Switch:** This is the ShoreTel switch associated with the night bell extension. The night bell extension can share the same switch port as the paging extension.
- **Paging Extension:** This is the extension used for your overhead paging system. This extension must be associated with a ShoreTel switch audio output port that you specify as the next parameter. There is only one paging extension per site.

  You must assign switches to the site and select the switch that will support the paging extension before you can save a paging extension.
- **Paging Switch:** This is the ShoreTel switch associated with the paging extension. The paging extension can share the same switch port as the night bell extension.
- **Operator Extension** This is the extension to which the user is transferred when dialing the operator digit for the site (typically "0").

  You must configure the appropriate user before assigning the operator extension.

This extension must not be confused with the Personal Assistant extension defined in the user's personal options. The Personal Assistant lets the user define the destination to which the caller is transferred when dialing "0" from the user's voice mail prompt. This might be an administrative assistant or a colleague, rather than the operator.

- **FaxRedirect Extension:** When a fax tone is detected, incoming calls are automatically transferred to this extension. Each site may have its own fax redirection number. Which fax redirection number is used depends on how the call is answered.
  - **If the user answers the call**, the fax redirection extension of the user's site will be used.
  - **If the call is answered by voice mail**, the Auto-Attendant or other menu, or a workgroup's queue step menu, the fax redirection extension at the site where the call originated is used. This is the site with the trunk that handled the inbound external call.

  The fax redirection extension must be an existing user.

Bandwidth parameters:

- **Admission Control Bandwidth:** This defines the bandwidth that voice streams can consume between the local site and all other sites. The caller hears a "network busy" prompt if this value is exceeded. To compute the admission control value for the site, see Chapter 9, "Network Requirements and Preparation" in the ***ShoreTel 12: Planning and Installation Guide***.
- **Intra-Site Calls** (calls within a site): This drop-down list has the types of encoding available for making calls within a site.
- **Inter-Site Calls (calls between sites):** This drop-down list has the types of encoding used for calls between ShoreTel sites.
- **Fax and Modem Calls:** This drop-down list has the types of encoding used for faxing or for calls made from a modem.

SIP Proxy parameters: SIP Proxy parameters support the ShoreTel SIP extensions. Refer to Section 18.1 for more information about SIP network elements.

- **Virtual IP Address:** This parameter defines the IP address of the site's SIP Proxy Server and Registrar server. The IP address is independent of the switch that performs the server functions. SIP extensions require that this parameter is set to a valid address.
- **Proxy Switch 1:** This setting designates the switch that performs the site's SIP server functions. The drop down menu lists all switches assigned to the site. SIP extensions require a setting for this parameter.
- **Proxy Switch 2:** This setting designates the switch that performs the site's SIP server functions when the switch specified by Proxy Switch 1 is not available. This parameter is optional.
- **Emergency Number List:** This is the list of numbers that can be dialed at the site with or without a trunk access code for emergency services. Note that this number must not conflict with any extensions.
  - **Trunk Access Code Required –** When this checkbox is selected, a caller must dial the Trunk Access Code before dialing the specified emergency number. If not selected, entering the Trunk Access Code before the Emergency number is permitted, but not required, to complete the call.

— **Data Entry Field –** Enter the exact emergency number required to contact the associated Emergency Service Provider. If Trunk Access Code Required is selected, you can also enter a number in canonical format.

— **Add More –** Click this button to create addition data entry fields for entering additional emergency numbers. Each site is permitted to have a maximum of ten emergency numbers to accommodate locations where multiple emergency service numbers are required.

- **Edit IP Phone Address Map:** This link opens the IP Phone Address Map Info edit page where you set the IP address range for sites other than Headquarters (see Figure 3-3).

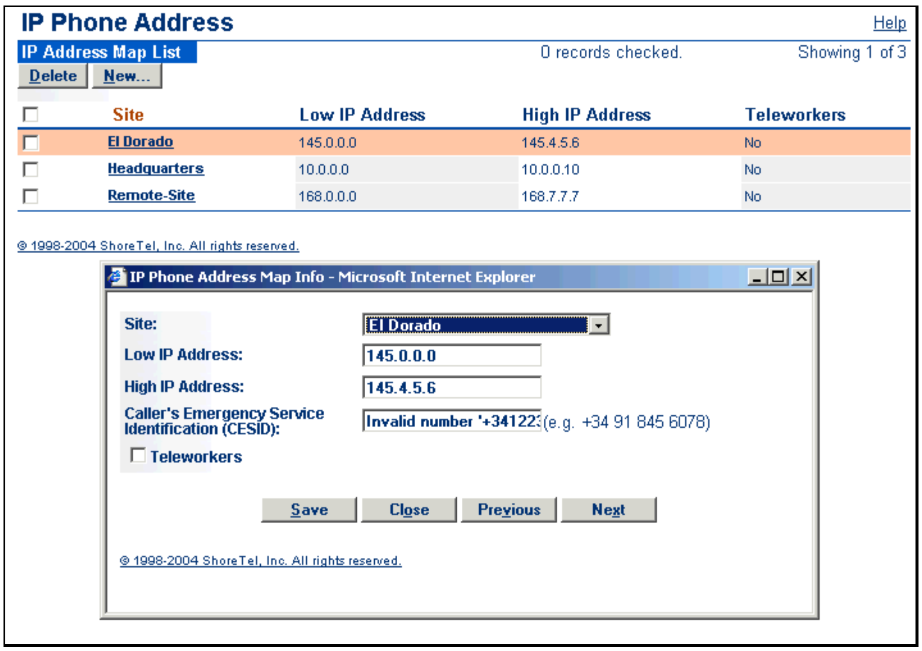


Figure 3-3 IP Phone Address Map Edit Page

All IP phones are assigned to Headquarters by default. If Headquarters is your only site, you do not need to set IP address ranges. If you have more than one site with IP phones, you must set an IP address range for each site (other than Headquarters).

This page is also accessible from the IP Phone Address edit page. For more information, see Chapter 8.

- **Site:** If you are setting the IP address range for a site other than the one shown in the Site drop-down list, select it from the list.
- **Low IP Address:** This is the lowest IP address in the range of addresses.
- **High IP Address:** This is the highest IP address in the range of addresses.
- **Caller's Emergency Service Identification (CESID):** Enter the Caller's Emergency Service ID to be used for IP phones in this IP address range. Enter, for example, +1 (408) 555-5555. This is the telephone number sent to the service provider when a user dials an emergency services number (e.g., 911 in the U.S.) and does not have a DID number or is in a user group for which the DID number is not to be used as CESID. This feature is only applicable to ISDN PRI trunks.

- **Teleworkers:** This call is an intersite call by the use of intersite codecs. The receiving site adjusts the bandwidth of the teleworker's call (at the receiving end).

CHAPTER 4

Configuring Application Servers

This chapter explains how to set up servers in a ShoreTel system. The topics discussed include:

4.1 Overview

The ShoreTel system supports not only distributed call control, but also distributed voice application servers. Distributed servers are extremely valuable for two purposes:

- Reducing WAN bandwidth by providing local voice mail and auto-attendant services
- Increasing the scale of the system

Even though there are multiple servers, the ShoreTel system provides a single image of your entire network. The system is currently certified to support up to 21 servers, one main server and up to 20 distributed servers. Consider adding a server at a site when the site exceeds 100 users. Add a new server for every 1,000 users.

The distributed servers run the following voice applications:

- **Voice Mail –** Each server supports 254 simultaneous voice mail or auto-attendant connections. The voice mail system uses SMTP to transport composed messages between the distributed servers. The ShoreTel system also supports linking to legacy voice mail systems using AMIS protocols.
- **Auto-Attendant –** The system supports up to 1000 menus that are hosted on every server, and each server provides 64 voice mail/auto-attendant connections.
- **Configuration –** The system enables users to log in and make configuration changes (call handling modes, etc.) from their ShoreTel Communicators client or from the Communicator for Mobile call handling mode client (if supported).
- **Maintenance –** The system provides a web site accessible through ShoreTel Director for maintenance of all the remote servers.

The distributed voice applications use a Remote TAPI Service Provider that relies on the call control information from the main server. Using redundant network paths to the main server can improve reliability of the remote server.

4.2 Distributed Voice Mail

The ShoreTel system uses Distributed Voice Mail (DVM) to provide greater voice mail availability. Each ShoreTel remote server has an instance of the telephony platform included, allowing voice mail and auto-attendant services to maintain full functionality during WAN short-term outages. The enhanced DVM included with the ShoreTel remote server allows users with mailboxes on the remote server to receive and pickup voice mail messages without depending on WAN connectivity to the headquarters server. The message waiting indicator (MWI) lights will correctly update with or without WAN connectivity.

Additionally, incoming calls can still reach the automated attendant, access the dial-by-name directory, and reach their intended local party during a WAN outage. If a party cannot be reached directly due to a WAN outage and his or her call handling would send unanswered calls to voice mail, the call is handled by the local voice mail server. The caller hears a generic greeting including the intended party's recorded name and can leave a message. This message will be forwarded at a later time to the home voice mail server for the addressee via SMTP.

Similarly, the enhanced DVM provides greater Communicator availability during WAN outages. If the WAN loses connectivity, users will retain full Communicator functionality as long as there is a DVM server at the same site as the users, the users voice mailboxes are on that server, and the DVM server is managing the switch that manages the users' phones.

Although each voice mail server is autonomous in delivering voice services, it still must have connectivity to the configuration data stored on the headquarters server in order to make configuration changes. Specifically, users on an isolated remote server would not be able to change call handling modes or make other changes that require modification to the configuration data on the headquarters server.

4.2.1 IP Phone Limitations/Requirements

Connectivity is required between the phone and the switch that is controlling the phone (this will be referred to as "basic connectivity"). All aspects of the phone's operation are functional when this basic connectivity exists, with the following exceptions:

- **Directory feature:** In addition to basic connectivity, the directory feature also requires connectivity between the switch and a headquarters (HQ) server or distributed voice mail (DVM) server that controls that switch.
- **Options features, Changes to Call Handling Mode (CHM), Wrap-Up:** In addition to basic connectivity, these features require either connectivity between the switch and an HQ server or DVM server that controls that switch. In addition, if the aforementioned switch is a DVM server, connectivity is required between that server and the HQ server or Distributed Database (DDB) services must be enabled for the DVM. Further, connectivity between the DVM server and the HQ server is required for successful synchronization between the Replication Master and Slave databases.
- **Switch-to-switch extension monitoring:** This condition exists when a programmed button requires monitoring activity on an extension that is serviced by a different

switch than the one that controls the phone. For example, if switch A (the phone's switch) is controlled by server X, and switch B (the monitored extension's switch) is controlled by server Y., then servers X and Y may be a DVM or the HQ server. For proper functionality of the switch-to-switch extension monitoring, the following conditions must exist:

— Switch A must be able to talk to server X.
— Server X must be able to talk to server Y.
— Server Y must be able to talk to switch B.
— If X and Y are the same, connectivity is, of course, assumed to exist.

- Auxiliary information on incoming calls, such as trunk information and called workgroup (WG) information, requires connectivity between the switch and a headquarters (HQ) server or distributed voice mail (DVM) server that controls that switch.

4.2.2 Communicator Limitations/Requirements

- **Communicator**: Communicator utilizes two communications channels, TAPI and CSIS. TAPI is used to communicate with the server that manages the switch that manages the user's phone (regardless of whether the phone is an analog or IP phone). CSIS is used to communicate with the user's voice mail server. (These two servers are often the same device.) As long as the client can reach these two servers, Communicator is fully functional.
- **First-time Communicator users:** When a user logs into Communicator for the first time, CSIS and TAPI both communicate with the HQ server in order to find out which server they need to use. Thus, for first-time users, a connection is required between the client and the HQ server regardless of where their VM and extensions are being serviced.
- **Workgroup functionality**: If users are configured to have workgroup functionality, they can access the mailboxes of all workgroups to which they belong. This requires connectivity to the server(s) on which those mailboxes reside.

4.3 Legacy Voice Mail Integration

4.3.1 Integration through Simplified Message Desk Interface

ShoreTel integrates with legacy phone systems for customers who would like to have the freedom and flexibility to continue to use their legacy systems while migrating toward a newer IP telephony solution. The legacy system must continue to work flawlessly regardless of whether calls are traversing the ShoreTel PBX on their way to the legacy voice mail system, or they are traversing the legacy PBX on their way to ShoreTel voice mail.

To address these needs, ShoreTel uses the Simplified Message Desk Interface (SMDI) protocol. SMDI allows dissimilar voice mail and PBX systems to work together. The protocol evolved at a time when voice mail services and PBX services were provided by separate physical devices, and enabled the disparate devices to share information over an out-of-band serial cable connection.

There are two modes of operation with respect to integrating a ShoreTel system and a legacy system using SMDI:

- **External voice mail –** In this configuration, the legacy system provides voice mail services while the ShoreTel system acts as the PBX for users.
- **ShoreTel voice mail –** In this configuration, the ShoreTel system provides voice mail services while the legacy system acts as a the PBX for users.

Voice mail extension lengths for the legacy voice mail system may be different from the ShoreTel voice mail extension lengths. In this case, digit translation information is required. For more information on digit translation tables, see the "Digit Translation Tables" on page 39.

For more information about integration to legacy voice mail systems using SMDI, see the ***ShoreTel 12: Planning and Installation Guide***.

4.3.2 Integration through Q Signaling Protocols

ShoreTel supports integrating the ShoreTel Unified Communications solution with other PBX platforms (see Section 5.5 for details about configuring basic QSIG services), and, as of ShoreTel 11, the Q-Signaling protocol's (QSIG) supplemental services for call diversion and message waiting indication. This integration allows a voicemail system located on either side of the QSIG link to be used by other system administrators to configure a ShoreTel user for voicemail that is hosted on a legacy PBX system using the same QSIG trunks on the same system.

QSIG is a Common Channel Signaling (CCS) protocol that runs over the ISDN D-channel for signaling between nodes in a Private Integrated Services Network (PISN). QSIG supports call setup, call tear down, and transparency of features such as message waiting, camp-on, and callback.

ShoreTel 11 and later support both ECMA and ISO versions of QSIG.

4.3.2.1 Configuring ShoreTel Users for External Voicemail with QSIG

The process for configuring QSIG External Voice Mail involves the following activities:

- Configuring a QSIG Tie Trunk to integrate with the external system (refer to the Configuring Trunks chapter for details about configuring tie trunks.)
- Defining QSIG Server integration
- Configuring a User Group for use of external QSIG Voice Mail
- Creating an Extension-Only user in the User Group

Use the following steps to configure QSIG External Voice Mail:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Voice Mail > External Voice Mail** (QSIG) menu. The External Voice Mail Servers (QSIG) page appears as shown in Figure 4-1.

Figure 4-1 External Voice Mail QSIG Page

Step 3 Select a site and click on the name to display the External Voice Mail Edit page as shown in Figure 4-2.

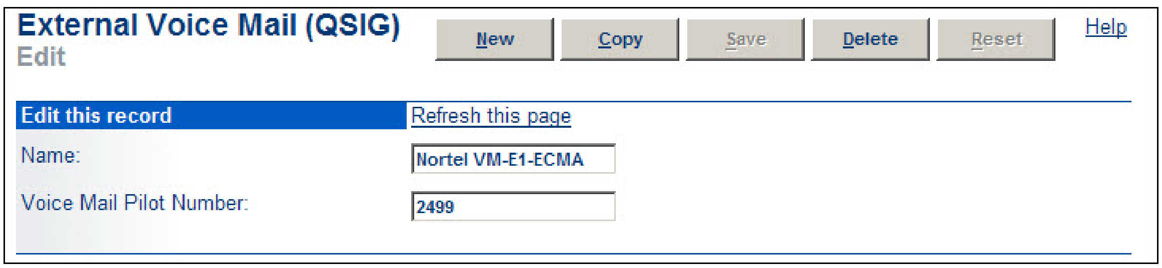


Figure 4-2 External Voice Mail QSIG Edit Record

Step 4 Type the name of the integration and the Pilot number for the voice mail. The pilot number is typically the OSE number for voicemail login or redirection.

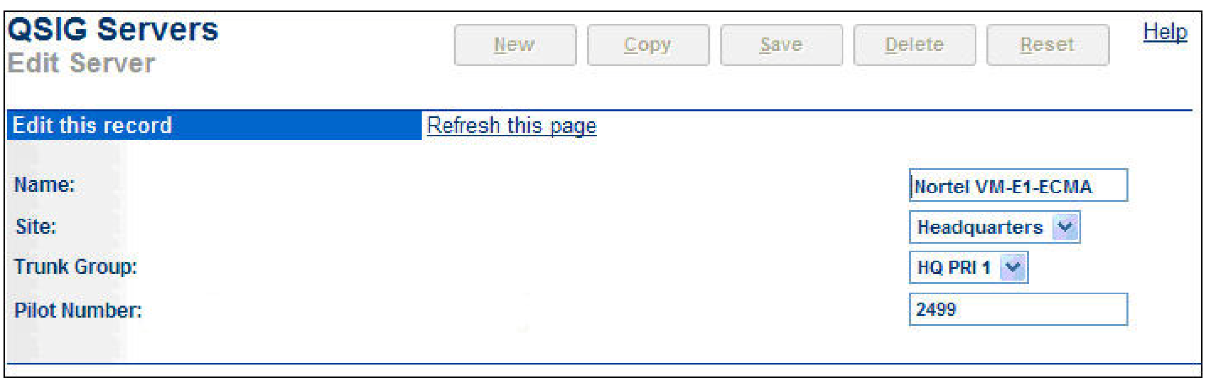


Figure 4-3 External Voice Mail QSIG Server Configuration

Step 5 Configure a User Group for using external QSIG voicemail by selecting External Voice Mail, QSIG in the drop-down scroll list for Voice Mail Interface Mode.

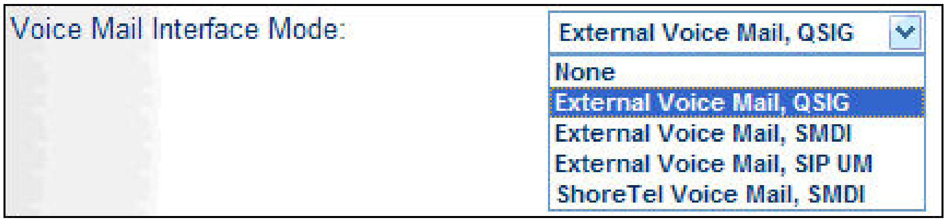


Figure 4-4 Voice Mail Interface Mode

See the ***ShoreTel 12: Planning and Installation Guide*** for sample Use Cases for implementing ShoreTel users with External Voice Mail QSIG.

4.3.2.2 Configuring Legacy Users for ShoreTel Voicemail via QSIG

The following steps describe the procedures necessary to configure an external user with ShoreTel voicemail service with QSIG External Voicemail.

Step 1 Launch ShoreWare Director.

Step 2 Click **Administration > Voice Mail > External Voice Mail** (QSIG). The External Voice Mail Servers page is displayed.

Step 3 Configure a Mailbox-Only account for the external user. The external user is now configured for ShoreTel voicemail on the ShoreTel system.

See the *ShoreTel Planning and Installation Guide* for sample use cases for implementing legacy users with ShoreTel Voice Mail QSIG.

4.3.2.3 Important Considerations

- The diversion implementation on both sides is not limited to voicemail service. Diversion due to call-forwarding, for example, is signaled using the same methods.
- ShoreTel features, such as "Find-Me," can result in multiple trunks being used to host a call.
  - No QSIG channel usage is available for secondary calls. Refer to ShoreTel's Norton Option 11C QSIG application note for more details on configuring this feature.

4.4 Configuring Application Servers

4.4.1 Adding Application Servers

The Application Servers > HQ/DVS link in the ShoreTel Director navigation frame provides access to the Application Servers list (Figure 4-5). The Application Servers list page is presented in alphabetical order.

By default, a ShoreTel system is configured with one server at the *Headquarters* site. Additional sites must have been configured before the steps in this section become available. For more information on adding remote sites, see Chapter 3: ShoreTel Sites, starting on page 65.

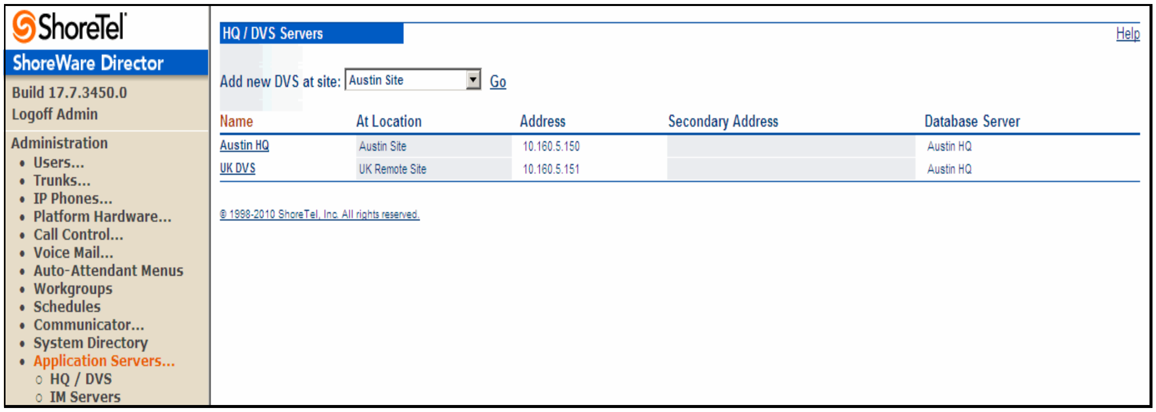


Figure 4-5 Application Servers List

To add a new server:

Step 1 In the “Add new application server at site” field, select the site.

Step 2 Click **Go**. The Application Servers edit page appears as shown in Figure 4-6.

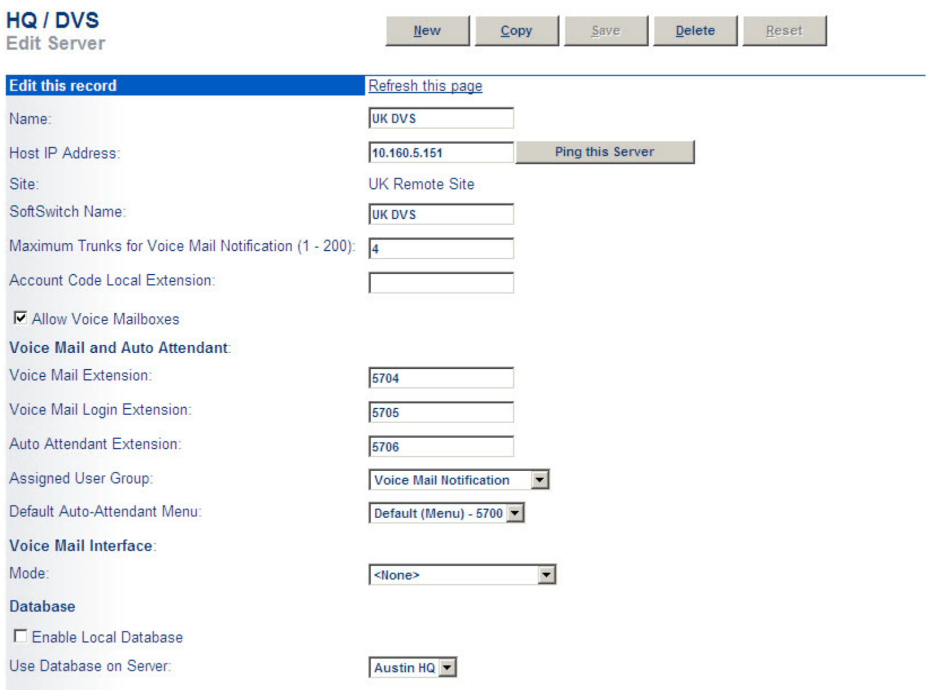


Figure 4-6 Application Servers Edit Page

Step 3 Enter parameters for the new server as described in the "Configuring Application Servers" on page 80.

Step 4 Click **Save**.

To edit an existing server:

Step 1 Select a server from the list in the Name column. The Application Servers edit page appears as shown in Figure 4-6 on page 79.

Step 2 Change parameters as needed for the new server.

Step 3 Click **Save**.

4.4.2 Configuring Application Servers

Figure 4-6 displays the Application Servers Edit page. The following is a list of parameters configured on this page:

Base Parameters

- **Name:** This is the name of a new or existing server.
- **Host IP Address:** This is the IP address of the server. If you enter the server host name, Director resolves the IP address when you click **Ping this Server**. You can also use the **Ping this Server** button to test the connectivity between your client PC and the new server.

  In system configurations that support Failover, this parameter specifies the Primary Server IP address. Refer to Section 20.3 for Failover information.
- **Secondary IP Address:** This is the IP address of the backup server that can assume controls when a system error results in a failover condition. Section 20.3describes ShoreTel system failover.
- **Site:** This is the physical location of the server. The location of the server is used to calculate bandwidth consumption for the purposes of admission control.
- **SoftSwitch Name:** This is the name of the SoftSwitch on the server you are editing. ShoreTel automatically creates a SoftSwitch for each server on the system.
- **Maximum Trunks for Voice Mail Notification (1 - 200):** This is the maximum number of trunks that can be used in the event of a voice mail notification. If many escalation profiles have been configured, it may be desirable to set this to a relatively low number to prevent notifications from overwhelming the system and making it impossible for users to make an outbound call.
- **Allow Voice Mailboxes:** Select this checkbox to allow voice mailboxes on this server. Clear this checkbox to prevent voice mailboxes to exist on this server.

  When the checkbox is disabled, voice mail configuration fields are still available because the server may still act as a backup VM server. However, SMDI is not available and the drop-down menu is disabled.

  If the server is being used as a VM server and mailboxes have been configured on that server, Director does not allow you to clear this checkbox.

  By default, this checkbox is selected.

Voice Mail and Auto-Attendant

- **Voice Mail Extension:** This extension is used by the voice mail server.
- **Voice Mail Login Extension:** This extension is used to log in to the voice mail server.
- **Auto-Attendant Extension:** This extension is used by the auto-attendant server.

- **Assigned User Group:** This is the assigned user group for the server. Because voice mail places outbound calls, the server must have assigned permissions.
- **Default Auto-Attendant Menu:** Each server can have a different default auto-attendant menu. This is the menu reached when none is specified - for instance, when a caller dials “9” to escape from voice mail and return to the auto-attendant.

Database

- Specifies whether the MySQL database is to be distributed and stored locally on the HQ system. See the "ShoreTel Distributed Database" on page 84 for information on ShoreTel Distributed Database service.

4.4.2.1 External Voice Mail Parameters

Figure 4-7 shows the Application Servers Edit page when Simplified Message Desk Interface is set to External Voice Mail. The configurable parameters in this window are described in this section.

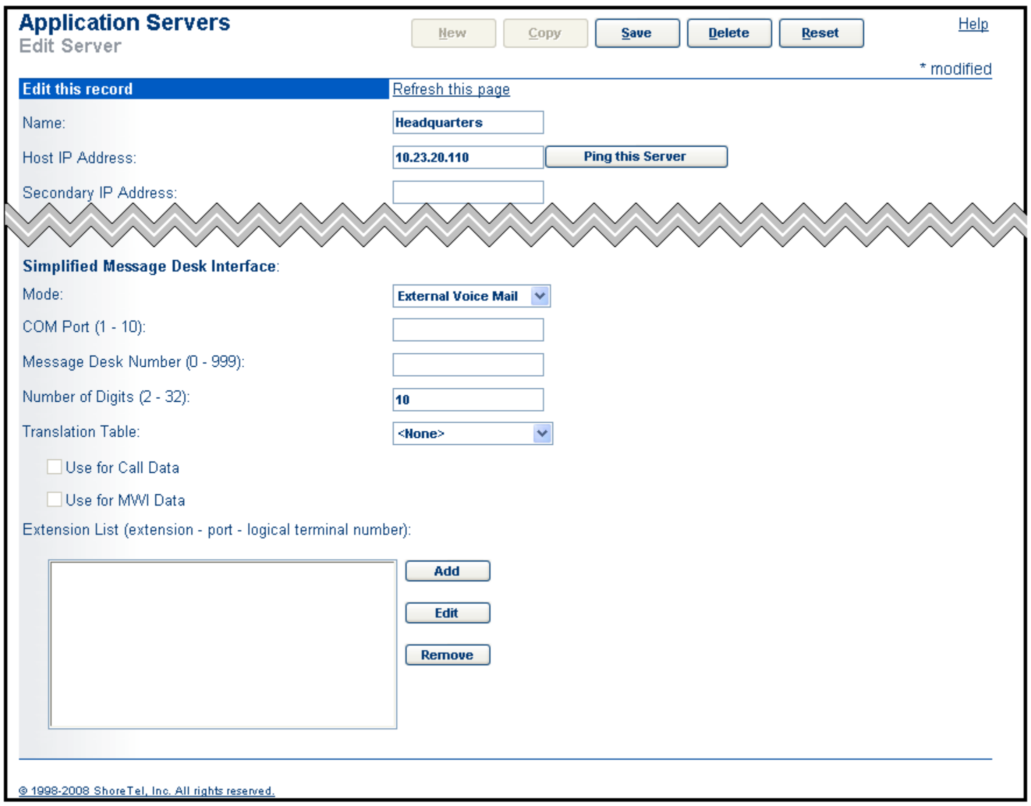


Figure 4-7 Application Servers Edit Page – SMDI set for External Voice Mail

- **External Voice Mail:** If this application server is to function as a PBX for a legacy voice mail system, check this box.
- **COM Port (1-10):** This is the COM port used by SMDI.
- **Message Desk Number (1-999):** The Message Desk default is 1. Valid values are 1 through 999. Set the number that the voice mail system expects. This parameter is most often set to one, since only one system will be using the SMDI link. In some configurations, however, a number of SMDI links can be daisy-chained together and

the Message Desk Number value is used to allow each system to know which data belongs to it.

- **Number of Digits (2-32):** This field sets the number of digits the ShoreTel system sends in the SMDI extension fields. Set this number to the value the voice mail system expects, most commonly 7 or 10. If the number of digits and the ShoreTel system extension value differ, the extension number is padded. For example, if ShoreTel needs to send extension 456 and the Number of Digits field is equal to 7, extension 0000456 is sent. If no padding is desired, the Number of Digits field would be set to 2 in this example. Then, only 456 is sent.
- **Translation Table Use for Call Data:** This check box indicates that the digit translation table is to be used for call data, when checked. Both Translation Table boxes may be checked at the same time.
- **Translation Table Use for MWI Data:** This check box indicates that the digit translation table is to be used for Message Waiting Indicator data, when checked. Both Translation Table boxes may be checked at the same time.
- **Extension List (extension - port - logical terminal number):** The SMDI message must contain the user extension, port number, and logical terminal number (exact trunk number). Note that these extensions forward to the Backup Auto-Attendant on No Answer or Busy.

4.4.2.2 External Voice Mail parameters

Figure 4-8 displays the Application Servers Edit page when Simplified Message Desk Interface is set to ShoreTel Voice Mail. The following parameters are configured on this page.

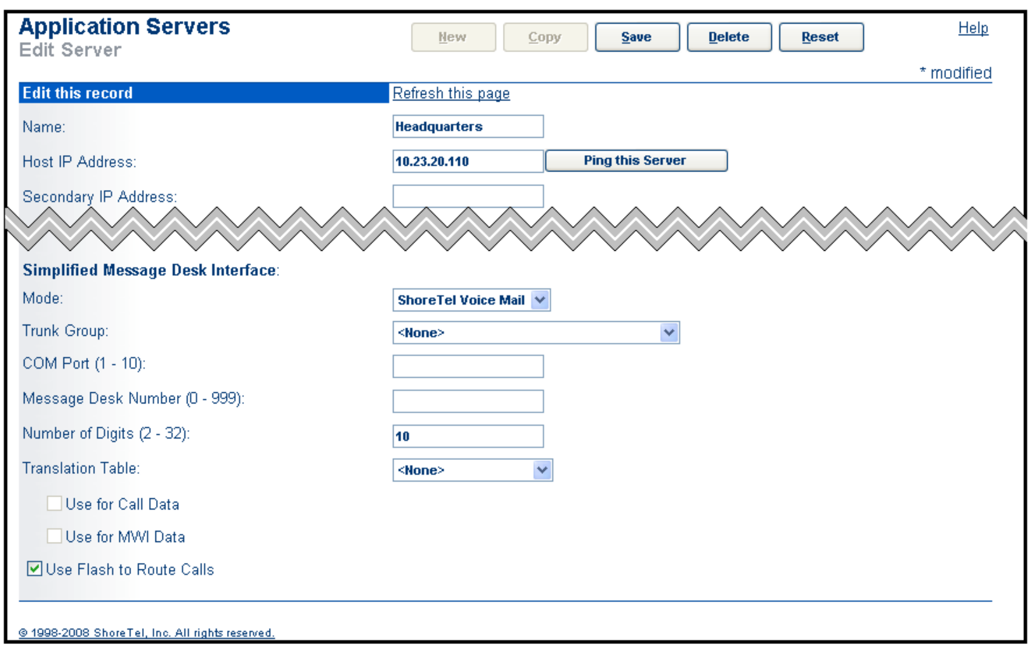


Figure 4-8 Application Servers Edit Page – SMDI set for ShoreTel Voice Mail

- **ShoreTel Voice Mail:** If this application server is to function as a voice mail server for a legacy PBX, check this box.

- **Trunk Group:** Select the trunk group to be used by the legacy PBX for voice mail traffic.
- **COM Port (1-10):** This is the COM port used by SMDI.
- **Message Desk Number (1-999):** The Message Desk default is 1. Valid values are 1 through 999. Set the number that the voice mail system expects. This parameter is most often set to one, since only one system will be using the SMDI link. In some configurations, however, a number of SMDI links can be daisy-chained together and the Message Desk Number value is used to allow each system to know which data belongs to it.
- **Number of Digits (2-32):** This field sets the number of digits the ShoreTel system sends in the SMDI extension fields. Set this number to the value the voice mail system expects, most commonly 7 or 10. If the number of digits and the ShoreTel system extension value differ, the extension number is padded. For example, if ShoreTel needs to send extension 456 and the Number of Digits field is equal to 7, extension 0000456 is sent. If no padding is desired, the Number of Digits field would be set to 2 in this example. Then, only 456 is sent.
- **Translation Table:** Select a translation table from the drop-down list. For information on creating translation tables, see “Digit Translation Tables” on page -39.
- **Use for Call Data:** This check box indicates that the digit translation table is to be used for call data, when checked. Both Translation Table boxes may be checked at the same time.
- **Use for MWI Data:** This check box indicates that the digit translation table is to be used for Message Waiting Indicator data, when checked. Both Translation Table boxes may be checked at the same time.
- **Use Flash to Route Calls:** Select this checkbox to use flash (i.e. a short hang-up to provide signaling instructions to a PBX) to route calls between the ShoreTel voicemail system and the legacy PBX. Enabling this feature may result in a more efficient trunk allocation.

  Note that analog trunks support the use of flash for this purpose, but other types of trunks (e.g. T1) do not.

  Clear this checkbox to prevent the system from attempting to use flash to route calls.

4.4.3 Extension List Mapping

To add extension list mapping to an application server configured for external voice mail, click Add found near the bottom of the Application Servers edit page. The External Voice Mail dialog box appears as shown in Figure 4-9.

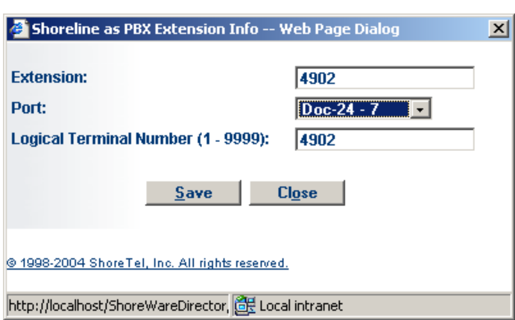


Figure 4-9 External Voice Mail Dialog Box

Enter the Extension to be used to access the legacy voice mail system. Also enter the physical Port to be assigned to the extension. And finally, include the Logical Terminal Number for the extension. Trunks in the trunk group that sends calls to external voice mail use a Logical Terminal Number. Make as many entries as are necessary.

For application servers configured for ShoreTel voice mail, select a translation table from the Translation Table drop-down list. For more information on creating a digit translation table, see "Digit Translation Tables" on page 39.

4.5 ShoreTel Distributed Database

4.5.1 Introduction

Organizations with remote locations often rely on the main or headquarters location to house and maintain the database for the entire company. Locating the database at the headquarters allows IT groups convenient access for upgrades and real-time maintenance activities. Organizations may also choose to distribute a read-only copy of the ShoreTel database on ShoreTel's Distributed Application Servers. Applications on these distributed servers (DVS), typically connect to a copy of the database running on the local server. However all write operations must still access the headquarters database. ShoreTel 11 introduces support for a distributed ShoreTel database that allows some actions previously requiring access to the HQ server. Prior to ShoreTel 11, users of ShoreTel Call Manager (now Communicator) could only change their call handling mode (CHM) if the HQ server was available. With ShoreTel 11 and later, user changes to their CHM are handled by the local ShoreTel server, even if the HQ server is not available. Using a distributed database also allows faster local queries and can reduce network traffic. Scalability is also improved because the headquarters server is no longer a bottleneck for database accesses.

4.5.2 Benefits of a Distributed Database

Availability – A remote server with DDB can run without disruption when HQ is down. Additionally, servers running a DDB can be rebooted and will operate properly, even when the HQ server is not available.

Scalability – Implementing a DDB on remote servers can reduce the workload on the HQ by handling queries locally, thus reducing the number of queries processed by the HQ system.

The following benefits are also realized by deploying a distributed database:

- ShoreTel Communicator users no longer need access to the HQ server in order to modify their CHM
- No action is normally required by the administrator after initial configuration The database on the HQ server acts as the replication master and remote
- servers are replication slaves.
- Updates from the remote server are automatically sent to the HQ database once it is available. All applications will continue working without changes while the HQ system is down and continue to function as the HQ database is updated.

4.5.3 Important Considerations and Warnings

Voice Mailbox Server switches (VMBs) do not support copies of the database.

Distributed workgroups and the distributed database features can not be configured on the same ShoreTel system. ShoreTel Director does not allow configuration of both of these features on the same ShoreTel system.

When DDB is activated, changing the name of the remote application server (DVM) will break the replication. A manual synchronization will be required to re-establish the DDB replication between the HQ and DVM servers.

4.5.4 Configuration of Distributed Database Service

4.5.4.1 Configuring the Replication Master and Slave

The ShoreTel installer will configure the HQ database as the replication master by enabling replication in for both new installs and upgrades.

The ShoreTel installer installs a MySQL instance on every remote server by default. However, this MySQL instance is not a writable copy of the ShoreTel database. All applications on the remote server will normally point back to HQ database by default. Specific configuration must be performed to enable applications to use the local copy of the MySQL database.

4.5.4.2 Configuring Distributed Application Servers to Host the ShoreTel Database

The ShoreTel “Application Servers” menu page contains a new section to configure the location of the database server as shown in Figure 4-10.

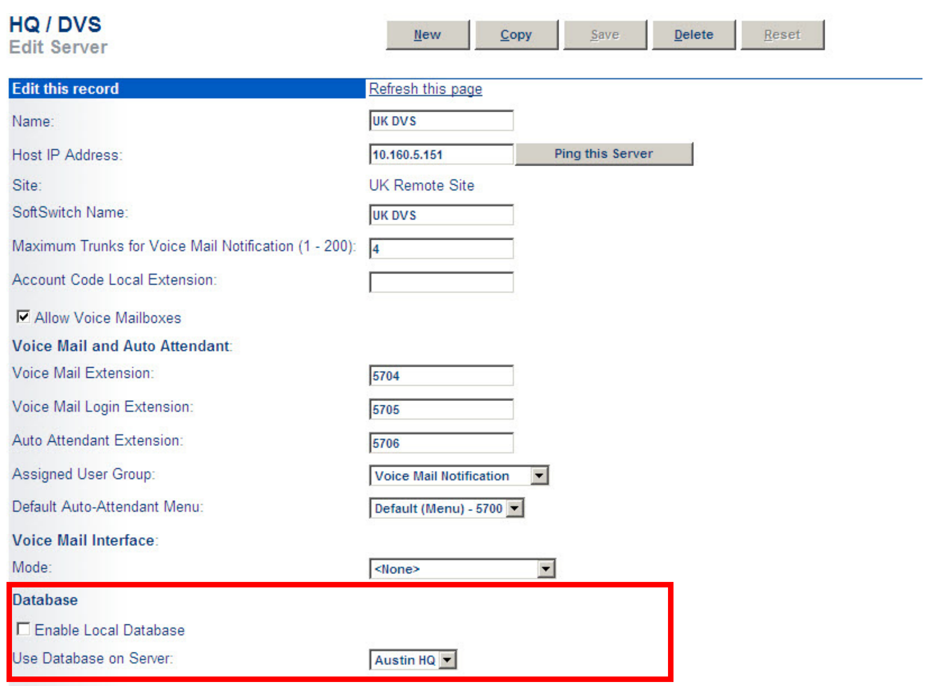


Figure 4-10Specifying the Distributed Database Usage

Administrators can create a local database by selecting the appropriate check box. The database instance can be created on the remote server as needed.

For example, a Distributed Voice Server can use the HQ database when initially installed. As the demands on the HQ server increase, the administrator may decide to add a local database instance on the DVS and configure the applications on the DVS to use the local database. ShoreTel provides a drop down list that will allow the DVS to switch to other databases, thus allowing for local or default database operation. For DVS's not configured with a local database, including VMBs, the administrator needs to select a proper database, usually a database server on the same site. Otherwise, the default action is to use the HQ database.

When the "Create Local Database" box is unchecked, the local database instance will be removed. If the local database is referenced by other DVS's, the operation will fail. If the DVS that hosts the database is the only one that references the database, deletion of the local database will be allowed. The DVS will then be switched to use the HQ database.

NOTE Distributed Voice Servers are not automatically setup to use a local database, even if another DVS on the same site has a local database configured. To use the database instance other than the HQ system, manual configuration is required.

4.5.4.3 Steps for Configuration of Replication Servers

After the Remote Application Server or DVS is configured to use a local database, the following steps need to be completed to start the replication process.

Step 1 Backup the database on the HQ system with the replication log position and transfer the dump file from the HQ system to the DVS system.

NOTE To reduce network traffic, it is recommended that the database dump file be compressed before transferring.

Step 2 Modify the my.ini file to enable the replication slave on the local DVS

Step 3 Restore the dump file on the DVS to create the local database instance and set the replication log position.

Step 4 Setup the replication master and start replication services using the SQL command.

Step 5 Restart the MySQL service to complete the configuration process.

NOTE These steps involve operations on both the HQ and the DVM systems and must be coordinated to ensure proper operation.

4.5.5 Distributed Database Status

4.5.5.1 DDB in QuickLook

ShoreTel Director Quicklook page show the database replication status in the Servers section. A small icon next to each server indicates whether the server has a database instance and uses color coding (Green or Red) to indicate the status of replication. To obtain details, click the server page to show more information.

4.5.5.2 DDB on Main Server Maintenance Menu

A new database section has been added to the main server maintenance page. It shows the master status, which include master log file name and master log position. Administrators can compare this information with the slave database information on the DVS maintenance page to determine how out-of-sync the remote instance may be with the HQ system. If connection to the HQ server is long-term, the systems may be synchronized manually. The synchronization point used is the last snapshot performed on the master database. The Snapshot button is used to generate an instant snapshot of the database which is used for synchronization or installation purposes.

4.6 Fax Servers

ShoreTel switches support direct connection to fax servers. Users can receive faxes sent to their voice number. When a call is answered (either by targeted user or through call forwarding) it is redirected to the fax redirection extension. The fax redirection extension is the first port allocated to the fax server. With multiple switch ports dedicated to the fax server, fax calls to user voice extensions are redirected to the first port connected to the fax server. If the first port is in a call, the fax is forwarded to the next port.

The ShoreTel switch sends original user's extension as DTMF immediately after call is answered. Fax server recognizes call is finished by loop current off. Once the fax call is complete, the fax server looks up user extension in its configuration. Fax server then routes the fax to correct end user. Depending upon the fax server's configuration, fax may be delivered as an email attachment.

For more information on fax server integration, see the ***ShoreTel 12: Planning and Installation Guide***.

CHAPTER 5

Configuring Switches

This chapter provides a general overview of the ShoreTel Voice Switches as well as information on how to configure them through ShoreTel Director. This chapter provides information on configuring ShoreTel Voice Switch:

The ShoreTel Voice Switches provide a highly reliable, highly scalable platform for the distributed call control software. Each ShoreTel Voice Switch connects to the IP network using a 10/100/1000M Ethernet port.

If more ports are required, you simply connect additional ShoreTel Voice Switches to your IP network. The system is inherently scalable, unlike legacy PBX systems that have hardware breakpoints with line cards, shelves, cabinets, and systems.

For more information about the features supported outside the U.S. and Canada, refer to the ***ShoreTel 12: Planning and Installation Guide***, Appendix A, "International Planning and Installation."

Topics discussed include:

5.1 Switch Models

ShoreTel Voice Switches are divided into two switch families that are based on chassis type:

- 1-*rack unit* (RU or just U) Half Width Switches (1 U is 1 3/4 inches)
- 1-U Full Width Switches

The sections that follow briefly introduce each switch family. Appendix C describes each ShoreTel Voice Switch. It includes LED behavior, interface details, capacity, and front panel illustrations.

5.1.1 ShoreTel 1-U Half-Width Voice Switches

The ShoreTel 1-U Half-Width Switch family is the most recent ShoreTel Voice Switch design. 1-U Half Width have a smaller footprint, use less power, and have lower heat dissipation requirements than earlier ShoreTel Voice Switches. These switches offer higher granularity in the number of IP users supported, allowing customers to precisely program the switch to satisfy their requirements.

The switches can be stacked or mounted in a standard 19-inch rack. Rack mounting 1-U Half Width Switches requires the ShoreTel Dual Tray. One or two switches are inserted into the Dual Tray, which is then mounted into the 19-inch rack. Two switches are mounted side by side Rack mounting the switches require the ShoreTel Dual Tray

ShoreTel 1-U Half Width Voice Switch models include:

- ShoreTel Voice Switch 30
- ShoreTel Voice Switch 50
- ShoreTel Voice Switch 90
- ShoreTel Voice Switch 90BRI
- ShoreTel Voice Switch 220T1
- ShoreTel Voice Switch 220T1A
- ShoreTel Voice Switch T1k
- ShoreTel Voice Switch 220E1

5.1.2 ShoreTel 1-U Full Width Voice Switches

The ShoreTel 1-U Full Width Switch family includes five models that support analog, IP, Session Initiation Protocol (SIP), T1, and E1 voice and data streams. Full width switch models can be stacked or mounted in a standard 19-inch equipment rack. These switches have a height of 1 U and an RJ21X connector for connection to analog phones and trunks. They also have redundant Ethernet LAN connections to ensure availability.

The ShoreTel 1-U Full Width Voice Switch models include:

- ShoreTel Voice Switch 120 – also called ShoreTel Voice Switch 120/24
- ShoreTel Voice Switch 60 – also called ShoreTel Voice Switch 60/12
- ShoreTel Voice Switch 40 – also called ShoreTel Voice Switch 40/8
- ShoreTel Voice Switch T1
- ShoreTel Voice Switch E1
- ShoreTel Voice Switch 24A

5.2 Switch Resources

ShoreTel Voice Switches provide telephony, IP phone, and SIP phone resources to ShoreTel users. Each voice switch offers a combination of resources that can be customized to support specific, individual configurations.

Forthcoming sections describe the resources available on ShoreTel Voice Switches. The description for each model enumerate the features available on the switch.

5.2.1 Analog Circuits

Voice switches offer three analog circuits: Extensions, DID trunks, and Loop Start trunks.

- **Extensions:** Extensions are telephony foreign exchange station (FXS) circuits that:
  - — Transmit and receive voice signals
  - — Supply power to phones
  - — Provide the ring signals and dial tone
  - — Indicate on-hook or off-hook state

  Director shows extensions as *analog ports*. They are assigned to user extensions.
- **DID Trunks:** DID trunks support inbound the Loop Reversal trunks that provide DID service from the central office. DID trunks are assigned to trunk groups.
- **Loop Start Trunks:** Loop start trunks are foreign exchange office (FXO) circuits that support inbound and outbound calls on ShoreTel IP Phones. These trunks accept ring signals, go on-hook and off-hook, and transmit and receive voice signals. Extensions (shown in Director as analog ports) are assigned to trunk groups.

  NOTE: Director windows show analog extensions as *ports*. For example, the heading under which extensions are displayed in Director is labeled "Port."

5.2.2 Digital Circuits

ShoreTel offers T1, E1, and BRI digital circuits that support Channel Associated Signaling (CAS) and Integrated Digital Service Network (ISDN) signaling through various 1-U Half Width and 1-U Full Width switches. Circuit channels are configured in Director Switch Edit windows.

5.2.3 IP Phone Ports

ShoreTel Voice Switches support varying numbers of ShoreTel IP Phones, as specified by the Switch's Edit panel. IP phone resources are allocated as follows:

- **Port Allocation:** Switch processing resources that support Digital and Analog ports on most ShoreTel Voice Switches can be reallocated to support five IP phone ports. For example, resources on a switch that supports 12 analog ports can be reallocated to support 60 IP phone ports.
- **Built-in Capacity:** Many switches provide processor resources that support IP phones without disabling telephony ports. Resources allocated to support IP phones cannot support SIP trunks or SIP proxies.

5.2.4 SIP Trunks

ShoreTel Voice Switches support varying numbers of SIP trunks, as specified by the Switch's Edit panel. SIP trunk resources are allocated as follows:

- **Port Allocation:** Switch processing resources that support Digital and Analog ports on most ShoreTel Voice Switches can be reallocated to support five SIP trunks. For example, resources on a switch that supports 12 analog ports can be reallocated to support 60 SIP trunks.

- **Built-in Capacity:** Many switches provide processor resources that support IP phones without disabling telephony ports. Resources allocated to support IP phones cannot support SIP Trunks or SIP Proxies.

5.2.5 SIP Proxies

ShoreTel switches support varying numbers of SIP trunks, as specified by the ShoreTel Director > Administration > Voice Switches/Service Appliances > Primary > Voice Switches page. SIP proxies resources are allocated as follows:

- **Port Allocation.** Switch processing resources that support Digital and Analog ports on most ShoreTel switches can be reallocated to support 100 SIP proxies. For example, resources on a switch that supports 12 analog ports can be reallocated to support 1200 SIP proxies.
- **Built-in Capacity.** Many switches provide processor resources that support IP phones without disabling telephony ports. Resources allocated to support SIP proxies cannot support SIP Trunks or SIP Proxies.

5.3 Configuration Parameters

Before configuring your switches in ShoreTel Director, you must determine the IP and MAC address assignments for each voice switch. Refer to the ***ShoreTel 12: Planning and Installation Guide*** for more information about getting an IP address for each voice switch.

The items that you need before you begin configuring your switches are:

- Name of each voice switch you are configuring.
- Internet Protocol (IP) address of each switch.
- Ethernet address (MAC address) of each switch.

The Ethernet address (MAC address) is printed on the ShoreTel voice switch rear panel.

5.3.1 IP Phone, SIP, and Make Me Conference Support

If you system is using IP phones, SIP devices, or SIP trunks you must allocate ports on ShoreTel Voice Switches. Each allocated port supports the following:

- Five IP phones
- Five SIP trunks
- 100 SIP devices

Make Me conference is used when a SIP trunk is involved in a conference call with three or more participants. All the Make Me conference settings are valid in this situation. Although only three parties are involved in a conference call involving a SIP trunk, four Make Me conference ports are reserved as this is an enforced rule for all Make Me calls.

If you do not reserve sufficient ports for IP phones on the switches, some or all IP phones will not be recognized or supported by the ShoreTel system. For more information in ShoreTel system requirements, see the ***ShoreTel 12: Planning and Installation Guide***.

5.3.2 Backup Operator

ShoreTel Voice Switches feature a backup operator in case the site operator is unreachable due to a network outage. For most switches, the backup operator is on the same port as the Power Fail Transfer port. To use this feature, select the port to match the switch model:

- Port 5 on the ShoreTel Voice Switch 40
- Port 9 on the ShoreTel Voice Switch 60 or the ShoreTel Voice Switch 120
- Port 12 on the ShoreTel Voice Switch 30, ShoreTel Voice Switch 50, ShoreTel Voice Switch 90, ShoreTel Voice Switch 220T1A

5.4 ShoreTel Director Windows for Voice Switches

Once a ShoreTel Voice Switch has been installed, its parameters can be configured through ShoreTel Director windows. This section describes the Director windows for configuring and monitoring ShoreTel Voice Switches. Subsequent sections provide details about the switch parameters.

5.4.1 Primary Voice Switches/Service Appliance

The Primary Voice Switches/Service Appliances page lists the ShoreTel switches installed in the ShoreTel network. To access the page:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Platform Hardware > Voice Switches/Service Appliances > Primary**. The Primary Voice Switches/Service Appliance page shown in Figure 5-1 appears.

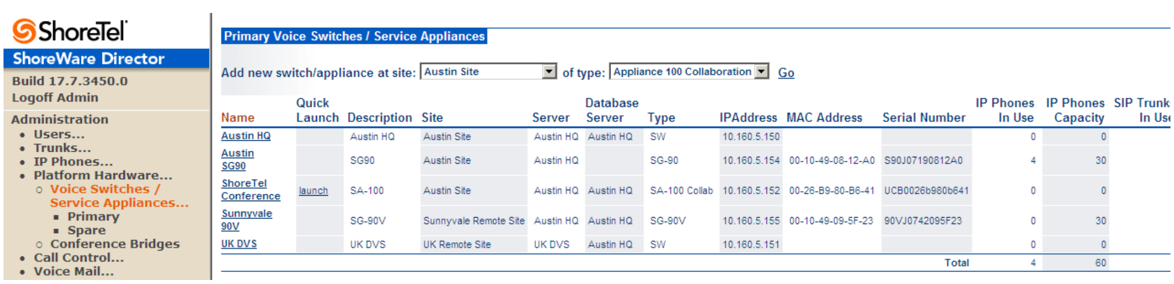


Figure 5-1 Primary Voice Switch/Service Appliance Page

5.4.1.1 Parameters in the Primary Voice Switch/Service Appliance Page

Each row corresponds to one switch installed in the ShoreTel network. Each column on the page lists a switch parameter as follows:

- **Name.** The name of the switch. Clicking a name takes you to the **Voice Switches** page, where you can modify the switch configuration.
- **Quick Launch. Provides a link that you can use** to open the Conference Administration page for the SA-100s installed on the system.
- **Description.** Describes the switch. It is an optional entry that typically tells where the switch is located or describes how it is used. For example, the switch's description might indicate the wiring closet where the switch is located.

- **Site.** The name of the site where the switch is located.
- **Server.** The name of server configured to manage the switch.
- **Database Server**. Shows the database server the device uses for backup.
- **Type.** The type of switch.
- **IP Address.** The switch's IP address.
- **MAC Address.** The switch's Ethernet MAC address.
- **Serial Number**. The serial number of the device.
- **IP Phones In Use.** The number of IP phones connected through the switch.
- **IP Phones Capacity.** The number of IP phones the switch can support based on the number of ports reserved for IP phones.
- **SIP Trunks In Use.** The number of SIP trunks connected through the switch.
- **SIP Trunks Capacity.** The number of SIP trunks the switch can support based on the number of ports reserved.
- **Conference Capacity.** The number of ports reserved for Make Me conferences.
- **Hunt Groups.** Shows the number of hunt groups the switch is hosting.
- **Music Source.** Indicates whether there is a music source for music-on-hold.

5.4.1.2 Adding a New Switch at a Site

To add a new switch at a site:

Step 1 Choose the site from the **Add new site/appliance at site** drop-down menu.

Step 2 Choose the type of switch from the **of type** drop-down menu.

Step 3 Click **Go**. The Voice Switches page for the specified switch is displayed.

5.4.2 Voice Switches Page

On the Voice Switches page, you can configure the identification and operating parameters of switches installed in the ShoreTel network. ShoreTel Director provides a specific page for each available ShoreTel switch to list only the relevant parameters for that switch.

The Voice Switches page typically consists of three sections:

- **Area to configure identification and signaling settings.** The list of parameters in this section depends on the type of switch being added to the ShoreTel network.
- **Switch graphic** (only on a Voice Switches page for existing switches) displays the switches front panel between the parameter section and the port table. The name of the switch is listed below the left side of the graphic. The port popout section provides port configuration information as follows:
  - — The color block specifies the port type assignment through the corresponding color blocks. Figure 5-3 provides the code for interpreting these blocks.
  - — Hovering the cursor on the LED graphic displays the Port Type setting and the Trunk Group, when appropriate, to which the port is assigned below the left side of the graphic.

The switch port graphical view appears at the bottom of the edit pages. The port, IP phone, Conference, SIP trunks, description, jack number, and location are displayed. Clicking a telephone or trunk port link takes you to the ***Port*** edit page for the associated port.

- Port table. Configures each port or channel on the switch.

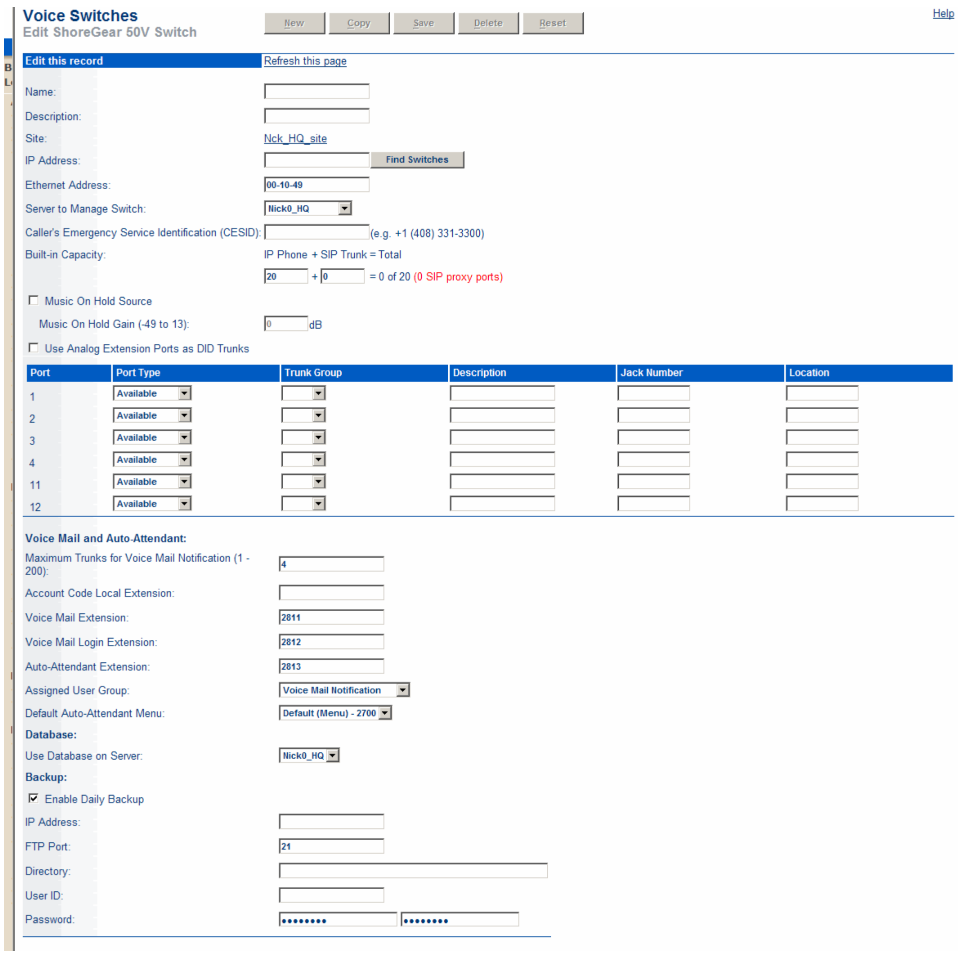


Figure 5-2 Voice Switches Page

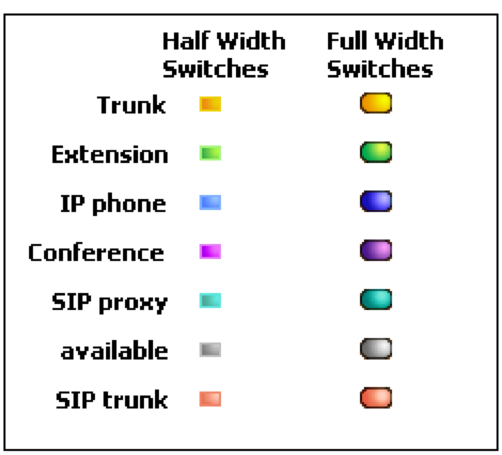


Figure 5-3 Port Type Assignment

5.4.2.1 Voice Switch Parameters

The Voice Switches page parameters depend upon and can vary with the model of switch that is being edited. These parameters are:

- **Name.** The name of the voice switch.
- **Description.** Describes the switch. It is an optional entry that typically tells where the switch is located or describes how it is used. For example, the switch's description might indicate the wiring closet where the switch is located.
- **Site.** Describes the site where the switch resides. This is a read-only parameter. If you want to move the switch to another site, you must move all the associated users and trunks, delete the switch from the current site, and add the switch to the new site.
- **IP Address.** The switch IP address.

If your DHCP/ BOOTP server is running, click **Find Switches** and use the resulting dialog box to select an IP address. This also adds the switch's MAC address in the **Ethernet Address** field. If the DHCP/ BOOTP server is not running, you must enter the switch's IP address and MAC address manually in the field.

- **Ethernet Address.** The switch's Ethernet address is the MAC address printed on the switch's back panel.

If the DHCP/ BOOTP server is running and you clicked **Find Switches** to select an IP address, the switch's MAC address was added at the same time in the **Ethernet Address** field. If the DHCP/ BOOTP server is not running, you must enter the switch MAC address manually in the field.

- **Server to Manage Switch.** The server that manages this switch. Select from the drop-down list.
- **Caller's Emergency Service Identification (CESID).** The telephone number sent to the service provider when an emergency services number is dialed from a user extension number. This parameter is not present on switches that do not contain ports that can be assigned to a user extension – SG T1, SG E1, SG T1k, and SG E1k.

Refer to "Emergency Dialing Operations" on page -595 for more information.

- **Built-in Capacity.** Built-in capacity allocates switch resources to support IP phones, SIP trunks, and SIP proxies on the ShoreTel network. Resource availability varies for each ShoreTel model.
  - — To allocate IP Phone and SIP Trunk resources, enter the desired number of resources in the data entry boxes.
  - — To determine the allocated SIP Proxy resources, subtract the number of available resources from the sum of the entered numbers, then multiply the difference by 20.

Example. The SG 90 provides 30 resources. If 5 resources are allocated for IP Phones and 5 resources are allocated for SIP trunks, then 400 SIP proxy resources are available: (30 - (5+5))*20.

- **Music On Hold Source.** Enables the music-on-hold port. Check this box to enable or disable this feature. This parameter enables and disables music on hold for all trunks, including SIP trunks, and cannot be applied to a specific trunk type.

Each site requires a separate music-on-hold source. Music is not available between sites across the WAN to save bandwidth. Enabling or disabling MOH for a switch only affects the local region associated with that switch. If MOH is enabled for a remote site but the headquarters switch has MOH disabled, then people calling into the headquarters switch will not hear music when placed on hold. Callers who dial into the remote site will, or course, hear music when placed on hold.

A music source, such as a CD player, must be connected to the Music On Hold jack on the front panel of the switch.

- **Use Analog Extension Ports as DID Trunks.** 1-U Half Width analog extension ports cannot be individually configured as DID trunks. Selecting this parameter configures all analog extensions as analog DID trunks. When this parameter is selected, analog ports on the switch cannot be assigned to a user extension port.

This parameter is not available on 1-U Full Width switches. Analog extensions these switches may be individually configured as Analog DID trunks from the Director Edit Trunk page.

T1 Signalling Parameters. Configures T1 circuit Layer 3 and Layer 1 parameters. These parameters are displayed for the SG T1, SG 220 T1, and SG 220T1A switches.

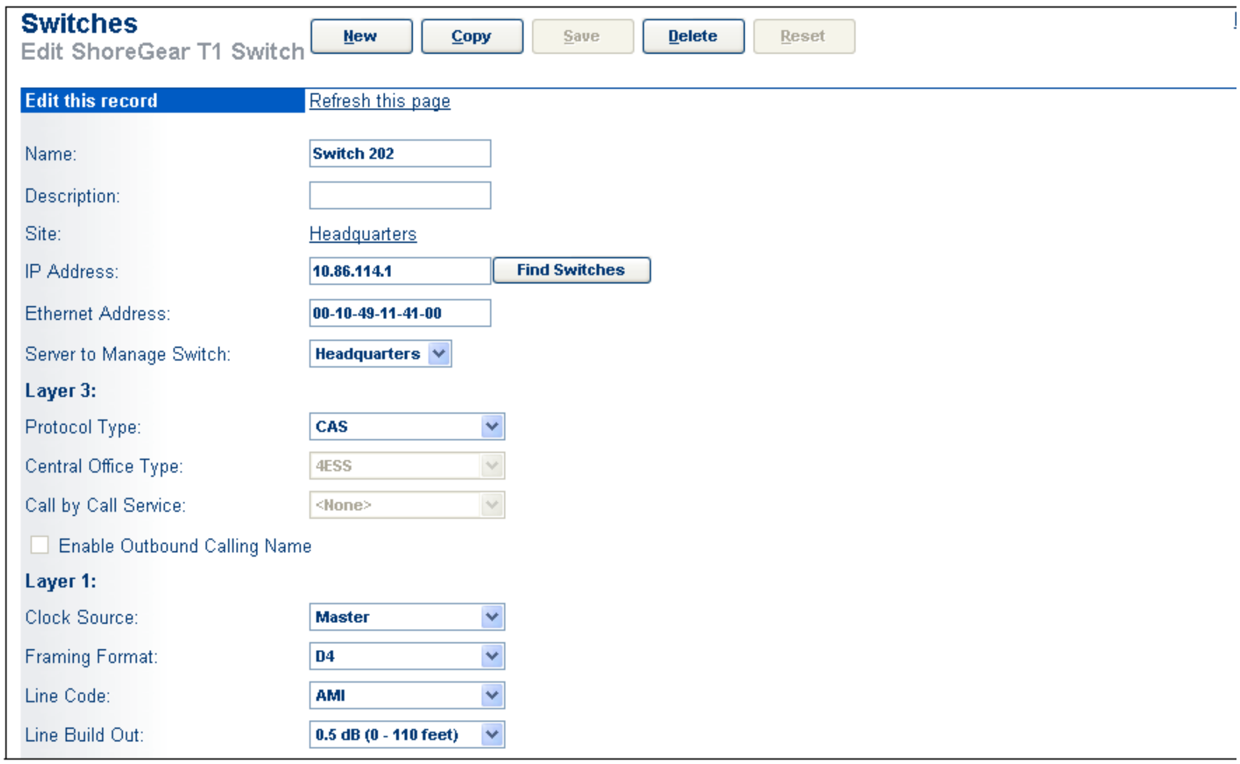


Figure 5-4 T1 Signalling Parameters

- **Layer 3.** Configures the network layer settings.
  - **Protocol Type:** Determines the type of signaling the ShoreTel-T1 will facilitate:
    - **CAS** uses in-band signaling where a portion of the bearer channel is used for A/B bit signaling, emulating on-hook and off-hook conditions.
    - **ISDN User** and **ISDN Network** is used in conjunction with ISDN signaling configurations. It provides a single D-channel for controlling 23 bearer channels. You can designate the T1 trunk as being **User** or **Network** side.

        * **QSIG Protocol** handles the signaling between digital PBXs for ISDN, handling the basic signaling functions of call setup, handshaking, and call teardown, and thus allowing PBXs from different third-party vendors to communicate and interoperate.
        * **QSIG Master** or **QSIG Slave**, depending on which type of signaling the ShoreTel-T1 will facilitate.

    Whereas PRI employs the concept of a "user" and a "network" to determine which network entity is controlling the D-link channel, QSIG uses a similar "master" and "slave" concept to determine the relationship between network elements.

- **Central Office Type:** Provides support for the Central Offices (COs). The default is 4ESS.
    - 4ESS
    - 5ESS
    - DMS-100
    - National ISDN-2 (NI-2)
- **Call by Call Service.** Call by Call Service is a 4ESS feature that allows a user to access different services, such as an 800 line or WATS line, on a per call basis. This parameter is available only when Central Office Type is set to 4ESS.

Parameter options (SDN or MEGACOM) are AT&T outbound service types.

- **Enable Outbound Calling Name.** Sends the caller name with the caller ID for outbound calls. The default is disabled.
- **Layer 1.** Configure the physical layer settings.
    - **Clock Source.** The ShoreTel-T1 switch's clock source. Depending on the type of T1 service provided by your telephone company's CO, select either **Slave** or **Master** from the drop-down list. Typically the ShoreTel-T1 is slave to the central office. The system default is **Slave**.
    - **Framing Format.** The ShoreTel-T1 switch's framing format. Depending on the type of T1 service provided by your telephone company's CO, select either **ESF** or **D4** from the drop-down menu. The system default is **ESF**.
    - **Line Code.** The ShoreTel-T1 switch's line code. Depending on the type of T1 service provided by your telephone company's CO, select either **B8ZS** or **AMI**. The system default is **B8ZS** from the drop-down menu.
    - **Line Build Out.** Provides a list of T1 trunk line distances, specified in decibels (dB) and in feet. Select the appropriate line code from the drop-down list.

E1 Signalling Parameters. configure E1 circuit Layer 3 and Layer 1 parameters. These parameters are displayed for the SG E1 and SG 220E1 switches.

- **Layer 3**. Configures the network layer settings.
    - **Protocol Type.** Specifies the signalling protocol.
        * ***ISDN User*** and ***ISDN Network*** are ISDN signalling protocols.
        * ***QSIG*** is an ISDN based signalling protocol used for signalling between PBXs in a private network.

— **Central Office Type.** The ShoreTel-E1 supports a single signaling type per country, which is typically **Euro-ISDN(TBR4)**. This parameter is active only if Protocol Type is set to **ISDN User** or **ISDN Network**.

— **Enable Outbound Calling Name.** When this parameter is selected, the caller name is sent with the caller ID for outbound calls. The parameter is disabled by default.

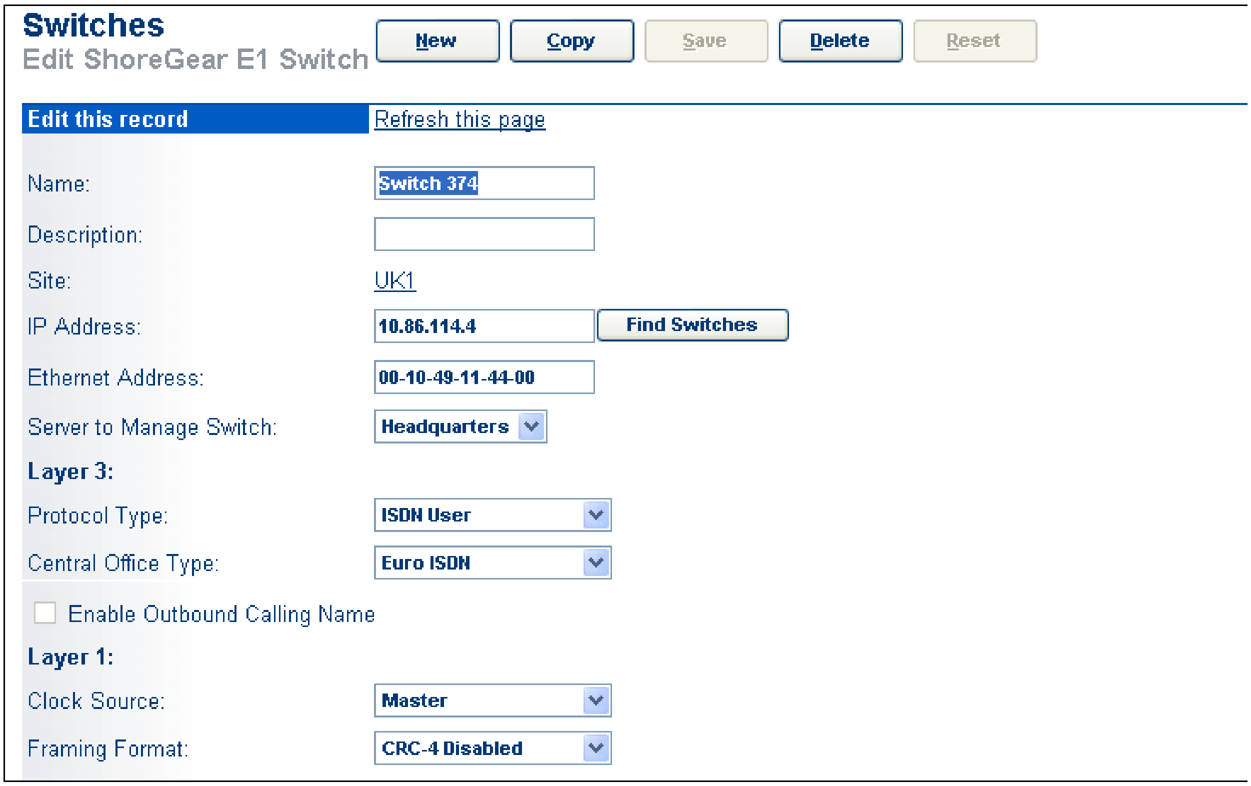


Figure 5-5 E1 Signalling Parameters

- **Layer 1.** Configures the physical layer settings.

  — **Clock Source.** Specifies the clock source. This parameter is typically set for compatibility with the service provided by the telephone company's CO. The default setting is **Slave**.

  — **Framing Format.** The ShoreTel-E1 supports **CRC-4**.

BRI Signalling Parameters. Configures BRI circuit Layer 1, Layer 2, and Layer 3 parameters. These parameters are displayed for the four BRI spans; the ShoreTel 30 BRI supports one BRI span.

- **Enable Span as BRI**. Activates the corresponding Digital Channels as a BRI span. When this parameter is not selected, the Channel resources are available for reallocation to support IP phones, SIP trunks, or SIP proxies.

- Layer 3
  - — **Protocol Type.** Specifies the signalling protocol.
    - ✻ **ISDN User** and **ISDN Network** are ISDN signalling protocols.
    - ✻ **QSIG** is an ISDN based signalling protocol used for signalling between PBXs in a private network.
  - — **Central Office Type.** The ShoreTel-E1 supports a single signaling type per country, which is typically **Euro-ISDN(TBR4)**. This parameter is active only if Protocol Type is set to **ISDN User** or **ISDN Network**.

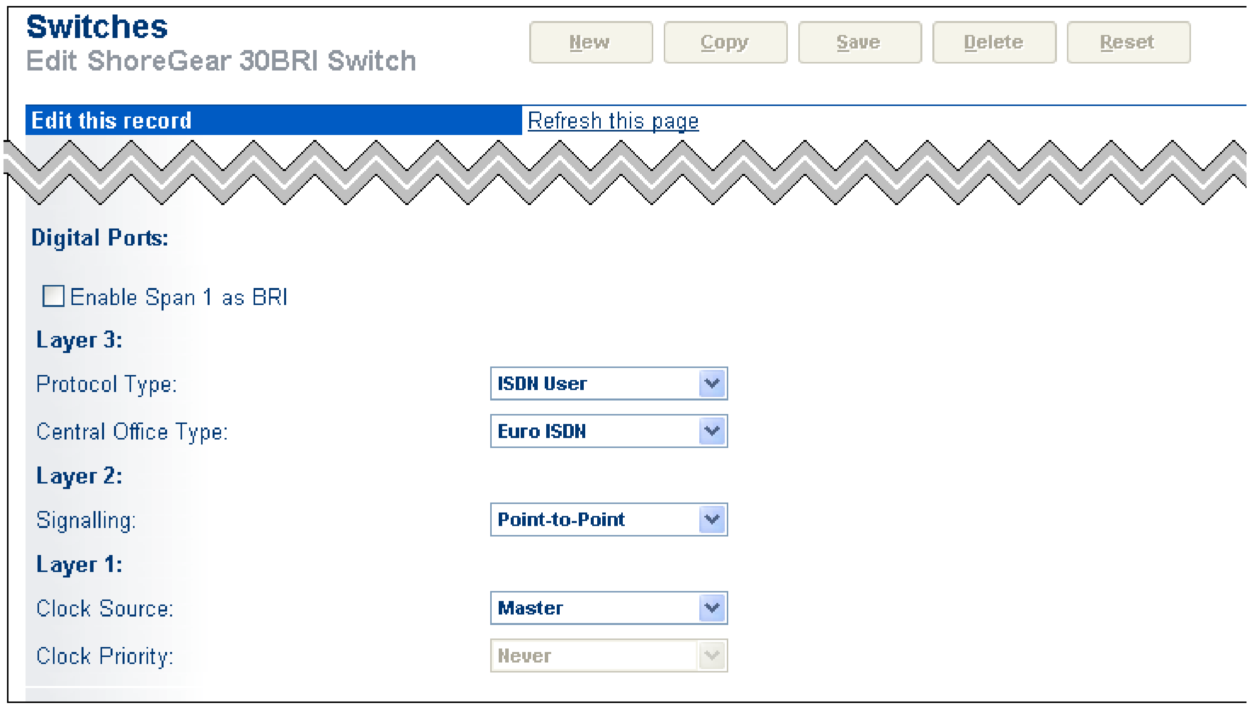


Figure 5-6 BRI Signalling Parameters

- Layer 2
  - — **Signaling.** Select either **Point-to-Point** or **Point-to-Multipoint**.
- Layer 1
  - — **Clock Source.** Specifies the clock source. This parameter is typically set for compatibility with the service provided by the telephone company's CO. The default setting is **Slave**.
  - — **Clock Priority.** Configures the clock recovery priority for the BRI voice channels.
    - ✻ When the Clock Source for the ShoreTel 90BRI is set to **Master**, the Clock Priority is set to **Never**.
    - ✻ When the Clock Source is set to **Slave**, this parameter configure the priority level: **High**, **Medium**, **Low**, or **Lower**. This parameter has no significance for the SG 30BRI, regardless of the Clock Source setting.

- **Analog Port Table**. For switches that provide analog circuit resources. Figure 5-7 displays an analog port table for an SG 90 switch.

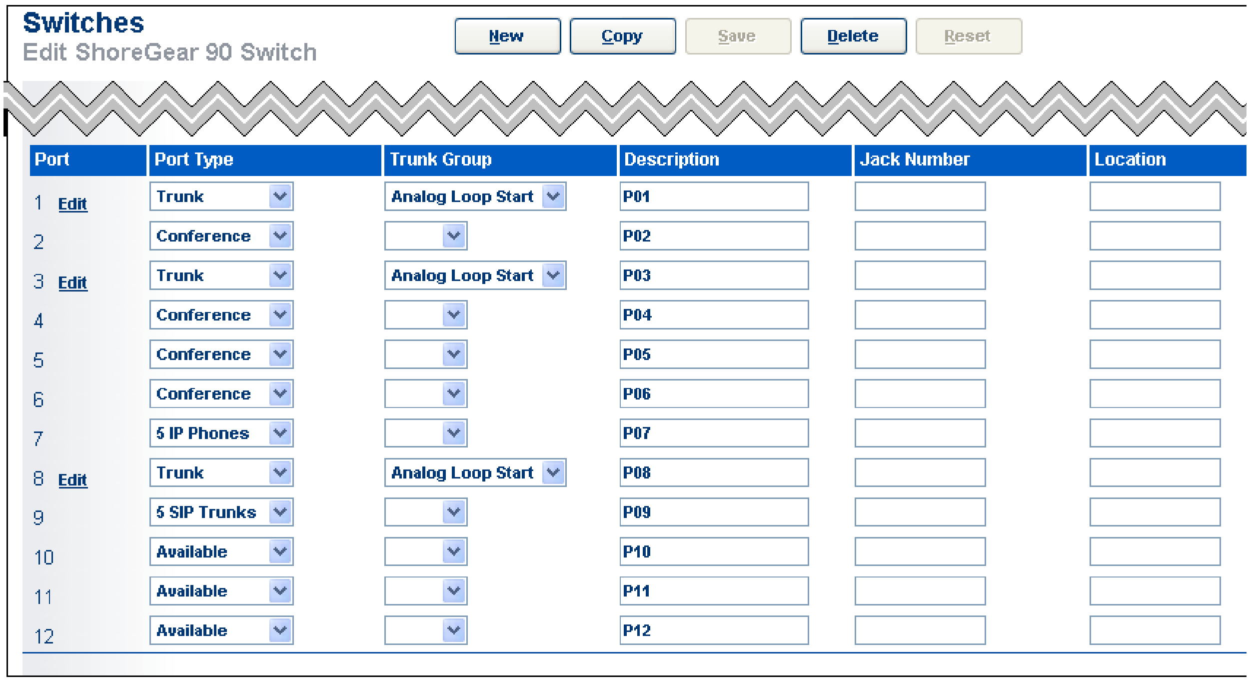


Figure 5-7 Analog Port Table

— **Port Type**. Configures the port resources.

- ⁕ **Available** configures the port resources to support either an extension port or DID trunk. Port capabilities depend on the ShoreTel switch model.
- ⁕ **Conference** configures the port resources for Make Me conferencing.

Reserve as many ports as you need to support the maximum number of conferences you will permit to occur simultaneously. For example, if you will allow two three-way calls at the same time, reserve 6 ports. The minimum number of ports that can be reserved for Make Me conferencing is four.

- ⁕ **Trunk** configures the port as an analog trunk assigned to the Trunk Group specified by the Trunk Group parameter.
- ⁕ **5 IP Phones** configures the resource to support 5 IP phones.
- ⁕ **5 SIP Trunks** configures the resource to support 5 SIP trunks.
- ⁕ **100 SIP Proxy** configures the resource to support 100 SIP proxies.

— **Trunk Group**. Specifies the Trunk Group to which the port is assigned. This parameter is available only when Port Type is set to Trunk.

Port tables do not include the Trunk Group column for switches that do provide Loop Start Trunks.

— **Description.** Optional field that lists a descriptive name for the switch port.

— **Jack Number.** Optional comment field for entering the patch-panel jack number to which the port is connected.

— **Location**. Optional comment field for entering location information about the port.

- **Digital Channel Table**. For switches that provide digital analog circuit resources.

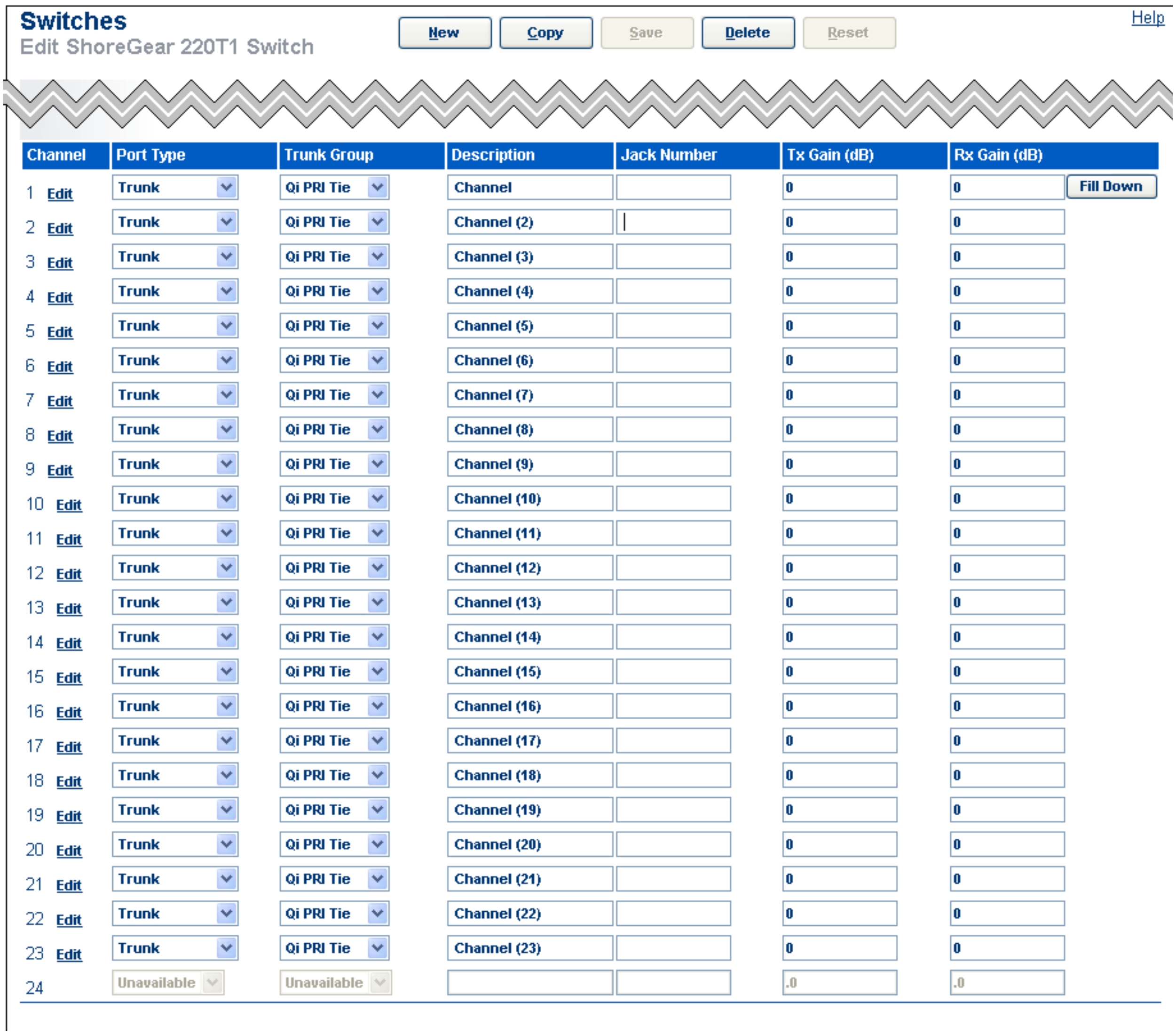


Figure 5-8 Digital Channel Table

— **Port Type.** Configures the port resources.

- **Available** indicates that the channel resources is available for assignment.
- **Trunk** configures the port as an digital trunk assigned to the Trunk Group specified by the Trunk Group parameter.
- **5 IP Phones** configures the resource to support 5 IP phones.

    - **5 SIP Trunks** configures the resource to support 5 SIP trunks.
    - **100 SIP Proxy** configures the resource to support 100 SIP proxies.
    - **Unavailable** indicates that digital channel is not available, such as in the case of a fractional T1 circuit.
  - **Trunk Group.** Specifies the Trunk Group to which the port is assigned. This parameter is available only when Port Type is set to Trunk.
  - **Description.** Optional comment field that lists a descriptive name for the switch port.
  - **Jack Number.** Optional comment field for entering the patch-panel jack number to which the port is connected.
  - **Tx Gain (db).** Specifies the gain added to received digital signals. The default is 0db.
  - **Rx Gain (db).** Specifies the gain added to transmitted digital signals. The default is 0db.
  - **Fill Down.** Duplicates first row data entry field contents in all other rows. The channel number, in parenthesis, is appended to the contents of the Description field.

5.5 Failover for IP Phones: Spare Switch

To ensure high availability of IP phones, ShoreTel provides a mechanism whereby IP phones maintain regular contact with most ShoreGear switches and contact the ShoreTel HQ server for immediate reassignment when the switch they are assigned to does not respond. When the system is properly setup and the IP phone failover function is enabled, IP phones send a pulse to the switch to which they are assigned every 60 seconds. If the switch fails to reply after four consecutive attempts, the IP phone sends a request to the HQ server for reassignment. If the HQ server determines that the switch is not available and other switches at the site can support additional phones, then the IP phones assigned to the down switch are redistributed among the remaining site switches on a first-come-first-serve basis. If resources are still insufficient and a spare switch is available for the site, the HQ server activates the spare server as a site resource and reassign to it the remaining IP phones. The HQ server records these failover transaction in such a way as to facilitate restoration. If HQ determines that resources are not available, then the phones are not reassigned to a switch and the user is unable to use it.

Failover for ShoreTel IP phones is invisible to the end user. A keep alive function means that failover can occur without users touching their phones and during active phone calls. If the user does try to use the phone before failover takes place, the phone automatically queries the HQ server for reassignment when the assigned switch does not respond. When implemented, the failover transaction occurs within seconds. However, if resources are not available, failover cannot occur and the user is unable to use the phone.

ShoreTel IP phone failover occurs on a first-come-first-server basis. If a switch goes down, the phones assigned to it are reassigned in the order that they contact the HQ server until all of the available resources are used. Provided the resources are available, the whole failover process from initial detection through failover transaction takes about four minutes.

The ShoreTel system provides for two levels of switch failover to assure high availability of IP phones. The first level involves setting aside capacity on site switches to handle failover situations. This method is referred to as N+1. In N+1 applications, you deploy more switches (hence ports) than your absolute need. The ShoreTel system automatically implements load balancing when it assigns IP phones to switches so that the load is always evenly distributed. An example of an N+1 application is the following:

> You have 99 users at a site. You deploy three ShoreGear 50 switches. The ShoreTel system assigns 33 IP phones to each switch leaving 17 ports in reserve on each switch. If one of the switches goes down, then the HQ server will reassign the 33 IP phones of the down switch to the two functioning switches without a hitch in the user experience.

The second level of failover involves installing a spare switch on the site for the purpose of providing failover protection. The switch is configure as a spare and the ShoreTel system does not assign IP phones to it under normal operating conditions. However, if a switch goes down and the HQ cannot reassign all of the IP phones to the remaining switches at the site, then the HQ server will activate the spare switch at the affected site and reassign to it remaining IP phones from the failed switch. However, the reassignment is intended to be temporary. The spare switch provides basic telephony functionality. You cannot configure the spare to provide such functions as hunt groups, trunk access, backup auto-attendant, analog extensions, trunks, media proxy, make-me conference ports etc. (Extension assignment are supported.)

Spare switch failover support is hierarchical. Spare switches provide failover support for IP phones installed on the same site only. Moreover, spare switches provide failover support for IP phones at or below the level where the switch is installed. This means that a switch installed on a child site cannot be used to provide failover for IP phones installed on the parent site or any site connecting through the parent site. It can be used to provide failover for child sites below it in the hierarchy. Indeed, when necessary, the system searches the entire, relevant hierarchy until it finds an available spare switch it can use for failover. You can also install spare switches on the HQ sites to provide universally accessible failover for all sites on the system.

This section describes how to set up and use spare switches in your system.

5.5.1 ShoreGear Switches that You Can Deploy as Spares

ShoreTel Voice Switches that can be configured as spares include:

- ShoreTel Voice Switch 120/24
- ShoreTel Voice Switch 40/8
- ShoreTel Voice Switch 60/12
- ShoreTel Voice Switch 50
- ShoreTel Voice Switch 90
- ShoreTel Voice Switch 220T1
- ShoreTel Voice Switch 220E1
- ShoreTel Voice Switch 90BRI
- ShoreTel Voice Switch 220T1A
- ShoreTel Voice Switch 30
- ShoreTel Voice Switch 30BRI

NOTE ShoreTel Voicemail switches cannot be used as spare switches.

NOTE Spare switches cannot change languages when they are activated. Language incompatibilities are denoted with an **Firmware upgrade available** message when configured into a site with a different language.

5.5.2 Physical Installation

The physical installation of a spare switch is the same as the installation of a primary switch. Refer to the documentation of the ShoreGear switch you want to deploy and the ***ShoreTel 12: Planning and Installation Guide*** for instructions about installing the switch on your network.

5.5.3 Adding a Spare Switch to the System

This section describes how to add a spare switch to the system.

Requirements

- IP address to assigned to switch.
- Ethernet address of switch (see label on the back of the switch).
- Name of the server that you want to use to manage the switch.

To configure a ShoreGear Switch as a Spare, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Platform > Voice Switches/Service Appliances > Spare**. The Spare Voice Switches page appears as shown in Figure 5-9.

Figure 5-9 Spare Voice Switches Page

Step 3 In the “Add new spare switch at site” field, select the site where you want to add the spare switch.

Step 4 In the “of type” field, select the switch model that you want to add as a spare.

Step 5 The Edit Switch page appears as shown in Figure 5-10.

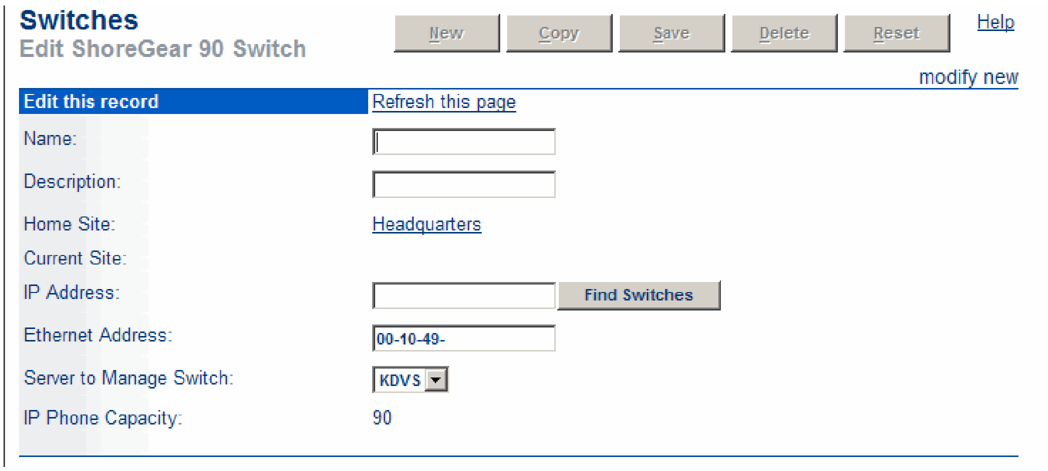


Figure 5-10Edit Switch Page

Step 6 In the Name field, enter the name that you want to use to identify this switch in the system.

Step 7 In the Description field, enter a description for this switch.

Step 8 In the IP Address field, enter the IP address assigned to the switch. If the switch is located on the same network segment as the HQ server, you can use the Find Switch function to locate the IP address.

Step 9 In the Ethernet Address field, enter the Ethernet address for the switch.

Step 10In the Server to Manage Switch field, select the server that you want to manage the switch.

NOTE We recommend that you do not select a server that has Music On Hold enabled to manage the spare switch. Doing so could mean sending MOH across the WAN which ShoreTel does not support.

NOTE We recommend that you do not select a server with CESID configured. The spare switch can be temporarily deployed in a remote location.

Step 11Click **Save**.

NOTE Home Site is the location where the switch is configured. Current Site is the site where the switch is currently being used to provide failover. If the switch is not currently being used, this field is empty.

5.5.4 Enabling IP Phone Failover

You must configure the system to allow IP phones to failover. To set the parameter to allow IP phone to failover, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > IP Phones > Options**. The IP Phone Options page appears as shown in Figure 5-11.

Figure 5-11IP Phone Options Page

Step 3 Check the **Enable IP Phone Failover** check box.

Step 4 Click **Save**.

5.5.5 Disabling IP Phone Failover Temporarily

There are times you when you will want to temporarily disable IP phone failover such system wide maintenance where switches are brought down. To disable IP phone failover temporarily, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Maintenance > Quick Look**. The Quick Look page appears as shown in Figure 5-16.

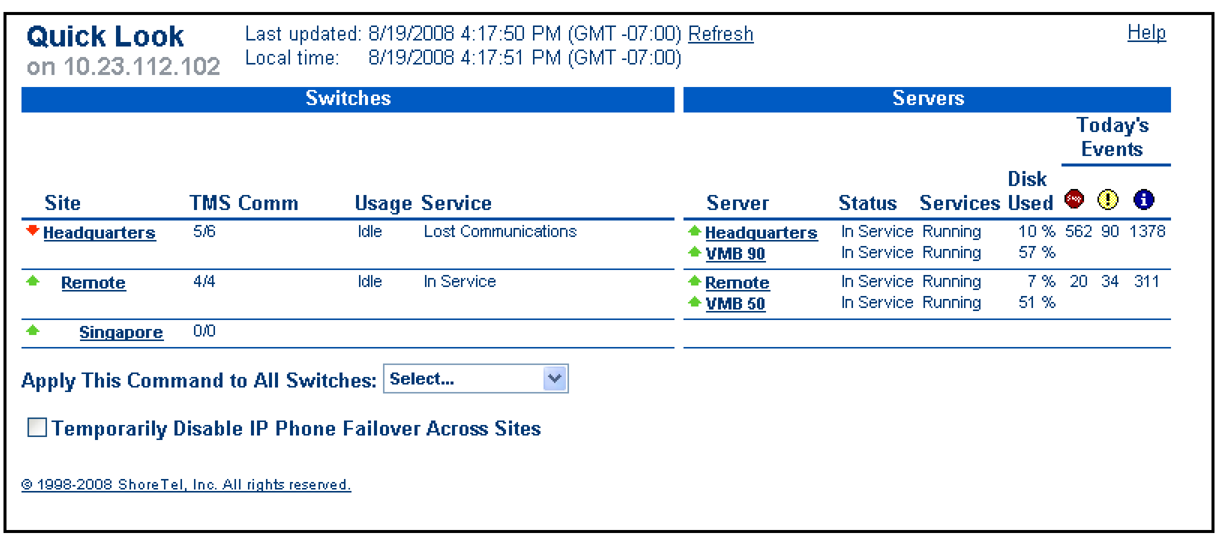


Figure 5-12Quick Look page

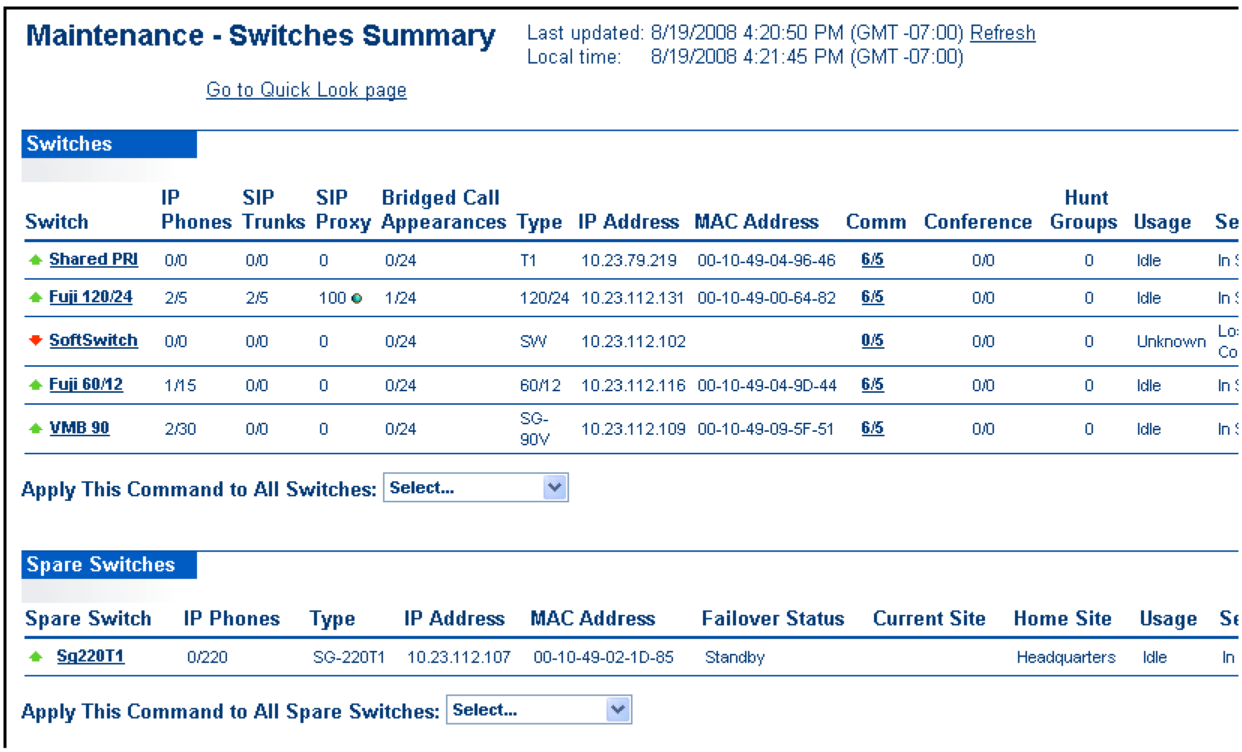
Step 3 Check the **Temporary Disable IP Phone Failover Across Sites** check box. When this option is selected, spare switches do not failover throughout the entire system.

NOTE Be sure to reverse this process to enable IP phone failover when your operation is complete.

5.5.5.1 Maintenance – Switches Summary

The Maintenance – Switches Summary page displays a section listing the Spare Switches located on the specified site. This section indicates the activity level of the switch and, when active, the site upon which the switch is deployed.

Manual failbacks are performed from this page by accessing the drop down menu for the desired switch and selecting **Fail Back**.



Maintenance - Switches Summary

Last updated: 8/19/2008 4:20:50 PM (GMT -07:00) Refresh
Local time: 8/19/2008 4:21:45 PM (GMT -07:00)

Go to Quick Look page

Switches

Switch	IP Phones	SIP Trunks	SIP Proxy	Bridged Call Appearances	Type	IP Address	MAC Address	Comm	Conference	Hunt Groups	Usage	Se
Shared PRI	0/0	0/0	0	0/24	T1	10.23.79.219	00-10-49-04-96-46	6/5	0/0	0	Idle	In S
Fuji 120/24	2/5	2/5	100	1/24	120/24	10.23.112.131	00-10-49-00-64-82	6/5	0/0	0	Idle	In S
SoftSwitch	0/0	0/0	0	0/24	SW	10.23.112.102		0/5	0/0	0	Unknown	Lo Co
Fuji 60/12	1/15	0/0	0	0/24	60/12	10.23.112.116	00-10-49-04-9D-44	6/5	0/0	0	Idle	In S
VMB 90	2/30	0/0	0	0/24	SG-90V	10.23.112.109	00-10-49-09-5F-51	6/5	0/0	0	Idle	In S

Apply This Command to All Switches: Select...

Spare Switches

Spare Switch	IP Phones	Type	IP Address	MAC Address	Failover Status	Current Site	Home Site	Usage	Se
Sg220T1	0/220	SG-220T1	10.23.112.107	00-10-49-02-1D-85	Standby		Headquarters	Idle	In

Apply This Command to All Spare Switches: Select...

Figure 5-13Switches Summary Page

5.5.6 Restoration

The spare switch is designed as a temporary measure to ensure that IP phone users have basic phone connectivity if their primary switch fails. To ensure that users have their full connectivity, you must repair or replace the failed primary switch as soon as possible. This section describes how to restore normal operation after a failover occurs. This section describes the following:

- How to re-assign the original primary switch profile to a new switch.
- How to move IP phones from the spare switch to the restored primary switch.
- How to failback the spare switch to the spare state.

5.5.6.1 Re-assigning the Primary Switch Profile to a Replacement Switch

If you must physically replace a primary switch that fails, you can re-assign the original switch profile to the new physical switch rather than create a new profile. This section describes how to re-assign the switch profile.

Requirements:

- Obtain a replacement switch that has the same capabilities as the failed switch.
- Physically install the replacement switch on the same network as the old switch.
- Assign the new switch an IP address. Refer to the x for more information about.
- Unplug the port connections (telephones, trunks) from the existing voice switch and plug them into the new voice switch.

To re-assign the switch profile, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Platform Hardware > Voice Switches/Service Appliances > Primary**. The Primary Voice Switches/Service Appliances page appears.

Step 3 Select the voice switch that is being replaced. The Edit Switch page for the switch appears.

Step 4 In the IP Address field, enter the IP address (or Ethernet address) of the new switch that you want to use to replace the downed switch.

NOTE You can also use the Find Switch button.

Step 5 Select the new voice switch and click **Save**. It can take up to two minutes for the switch to come on line.

NOTE You can use Quick Look to confirm that the new voice switch is on line.

5.5.6.2 Moving IP Phones to Primary Switch

To move IP phones from the spare switch to the primary switch, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > IP Phones > Individual IP Phones**. The IP Phones page appears as shown in Figure 5-14.

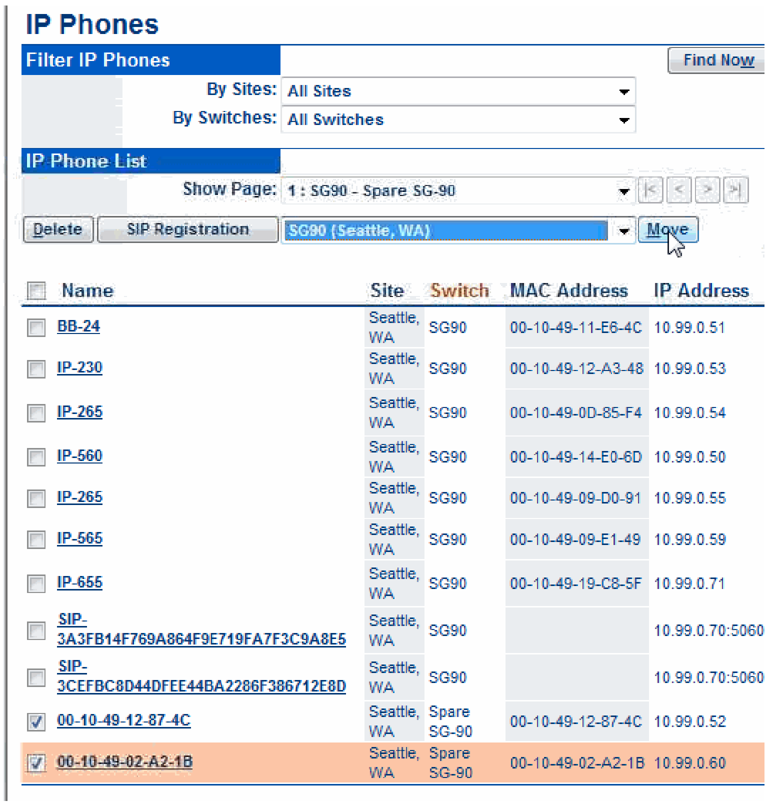


Figure 5-14IP Phone Page

Step 3 In the By Sites field, select the site where the failover has occurred and you want to perform restoration.

Step 4 In the Use the Switches field, select All Switches.

Step 5 In the Show Pages field, select the page that contains the IP phones that you want to move. The first name and the last name of the phones listed on the page are shown in the field along with the page number.

Step 6 Use the Site column to identify phones that have failed over to the spare switch that you want to move to the primary switch and check the check box to the left of the phone names (the MAC or IP address are often used as name).

NOTE You can select multiple phones to move at one time. The phones do not have to be registered to the same switch.

Step 7 In the field to the left of the Move button, select the switch to which you want to move the IP phones.

Step 8 Click **Move**. The phones are moved to the target primary switch.

NOTE Calls that are currently in progress are dropped during the move.

5.5.6.3 Failing Back the Spare Switch

After you move the IP phones to the primary switch on the site, you must manually failback the spare switch. To failback the spare switch, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Maintenance > Quick Look**. The Quick Look page appears.

Step 3 Select the site where the failover occurred and to which the spare switch is currently assigned. The Voice Switches and Service Appliances Summary page appears similar to Maintenance Summary page shown in Figure 5-15.

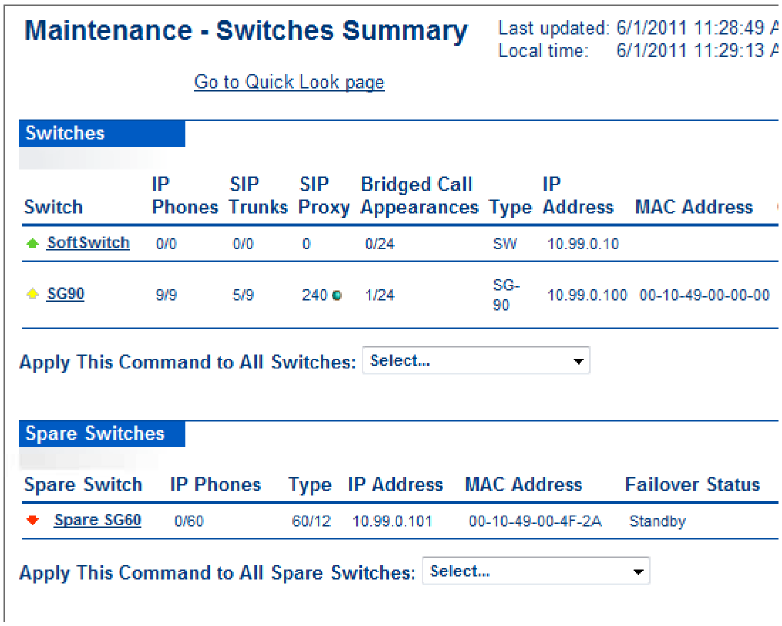


Figure 5-15Maintenance Summary Page

Step 4 In the Spare Switches section, identify the spare switch to fail back, and in the Command column field, select **Failback**. The failback process starts.

NOTE Make sure that there are zero (0) IP phones connected to the switch. (The listing in the IP Phones column should be 0/N, where 0 is the number of phone currently registered with the switch and N is the switch capacity.)

The process takes a few minutes to complete and includes rebooting the spare switch. When the process is complete and successful, the spare switch returns to the spare state. To verify that the switch has returned to the spare state, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Platform > Voice Switches/Service Appliances > Spare**. The Spare Voice Switches page appears.

Step 3 Verify the following columns:

- The Current Site column is empty.
- The IP Phones in Use column lists zero (0).

5.5.7 Disabling IP Phone Failover Temporarily

There are times you when you will want to temporarily disable IP phone failover such system wide maintenance where switches are brought down. To disable IP phone failover temporarily, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Maintenance > Quick Look**. The Quick Look page appears as shown in Figure 5-16.

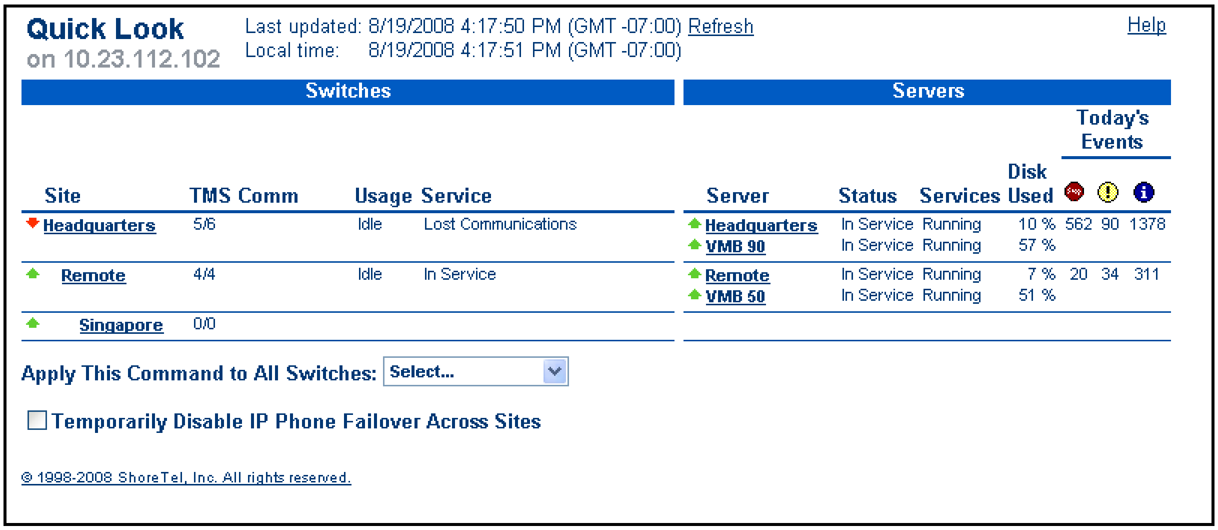


Figure 5-16Quick Look page

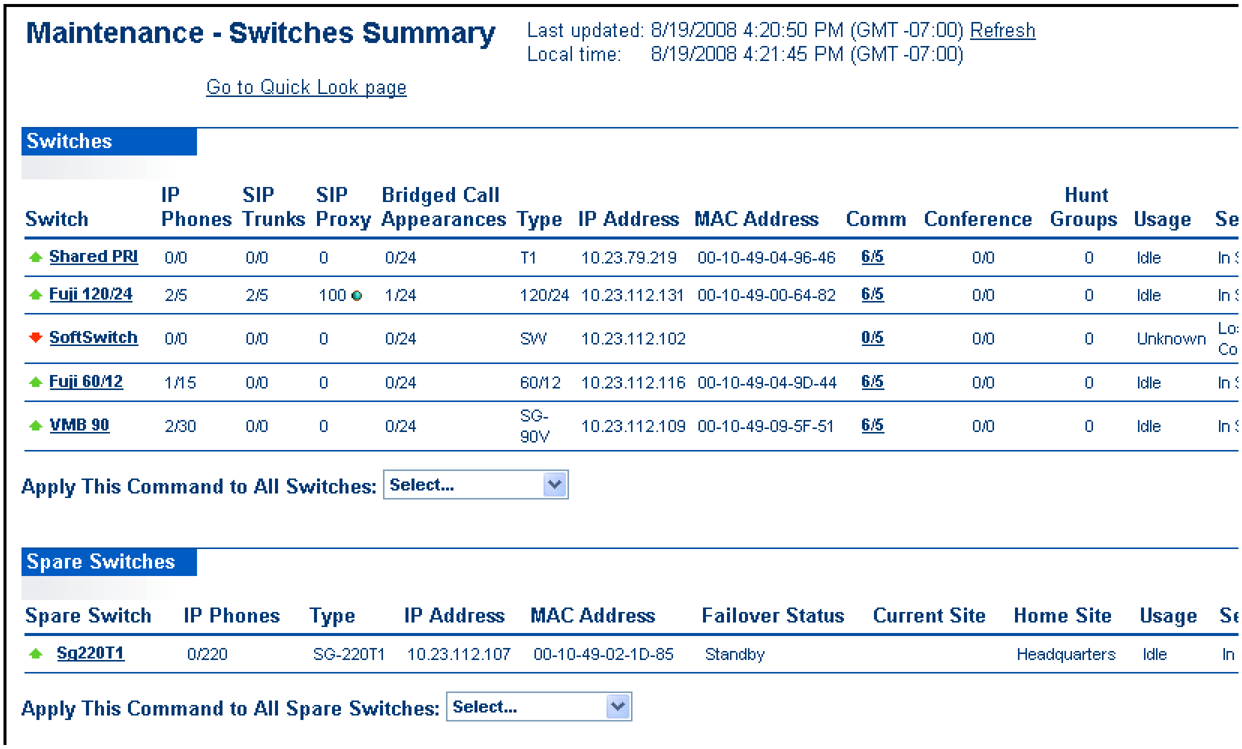
Step 3 Check the **Temporary Disable IP Phone Failover Across Sites** check box. When this option is selected, spare switches do not failover throughout the entire system.

NOTE Be sure to reverse this process to enable IP phone failover when your operation is complete.

5.5.7.1 Maintenance – Switches Summary

The Maintenance – Switches Summary page displays a section listing the Spare Switches located on the specified site. This section indicates the activity level of the switch and, when active, the site upon which the switch is deployed.

Manual failbacks are performed from this page by accessing the drop down menu for the desired switch and selecting **Fail Back**.



Maintenance - Switches Summary
Last updated: 8/19/2008 4:20:50 PM (GMT -07:00) Refresh
Local time: 8/19/2008 4:21:45 PM (GMT -07:00)
Go to Quick Look page

Switches

Switch	IP Phones	SIP Trunks	SIP Proxy	Bridged Call Appearances	Type	IP Address	MAC Address	Comm	Conference	Hunt Groups	Usage	Se
Shared PRI	0/0	0/0	0	0/24	T1	10.23.79.219	00-10-49-04-96-46	6/5	0/0	0	Idle	In S
Fuji 120/24	2/5	2/5	100	1/24	120/24	10.23.112.131	00-10-49-00-64-82	6/5	0/0	0	Idle	In S
SoftSwitch	0/0	0/0	0	0/24	SW	10.23.112.102		0/5	0/0	0	Unknown	Lo Co
Fuji 60/12	1/15	0/0	0	0/24	60/12	10.23.112.116	00-10-49-04-9D-44	6/5	0/0	0	Idle	In S
VMB 90	2/30	0/0	0	0/24	SG-90V	10.23.112.109	00-10-49-09-5F-51	6/5	0/0	0	Idle	In S

Apply This Command to All Switches: Select...

Spare Switches

Spare Switch	IP Phones	Type	IP Address	MAC Address	Failover Status	Current Site	Home Site	Usage	Se
Sg220T1	0/220	SG-220T1	10.23.112.107	00-10-49-02-1D-85	Standby		Headquarters	Idle	In

Apply This Command to All Spare Switches: Select...

Figure 5-17Switches Summary Page

5.6 SoftSwitch

SoftSwitch is used to host virtual users who are not assigned a physical telephone port on any ShoreTel voice switch. The SoftSwitch is used to host all voice mail, auto-attendant, and workgroup extensions as well as route points. A SoftSwitch is automatically created for every server added to the ShoreTel system. The server will be listed as a SoftSwitch on the Switches list page.

To configure SoftSwitch, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Voice Switches/Service Appliances > Primary**. The Primary Voice Switches/Service Appliances page appears.

Step 3 In the Name column, select the SoftSwitch that you want to configure. You can use the Type column to identify softswitches. Softswitches are labeled SW in the Type column. The Edit SoftSwitch page appears as shown in Figure 5-18.

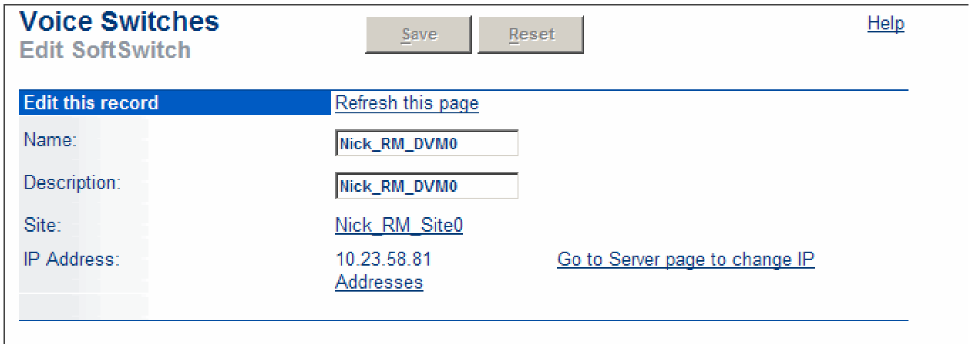


Figure 5-18Voice Switches Page for SoftSwitch

The parameters that appear on this page are:

- **Name.** The name of the SoftSwitch. The default name is SoftSwitch, but you can change the name to suit your system.
- **Description.** A descriptive name of the SoftSwitch.
- **Site.** The location of the SoftSwitch. This is a read-only parameter, and cannot be changed. The SoftSwitch is always located at the main site.

  The name of the main site is defaulted to "Headquarters," but can be changed via the **Edit Site** page.
- **IP Address.** This read-only field is the IP address of the switch supporting the SoftSwitch. Change the IP address from the **Server** page.

5.7 T.38 Support on ShoreTel Switches

Online FAXing has become a very popular trend for many enterprise organizations. The ability to quickly fax important documents or papers is vital to maintaining the type of business communications expected by most companies. Some ShoreTel switches support T.38 to provide improved reliability for fax calls. T.38 support is implemented as a fax codec on ShoreTel switches and performs as a gateway.

T.38 support is implemented on both ShoreSIP and SIP. Translation between SIP and ShoreSIP is implemented so that ShoreTel based switches can talk with SIP based T.38 switches. Figure 5-19 shows how T.38 works.

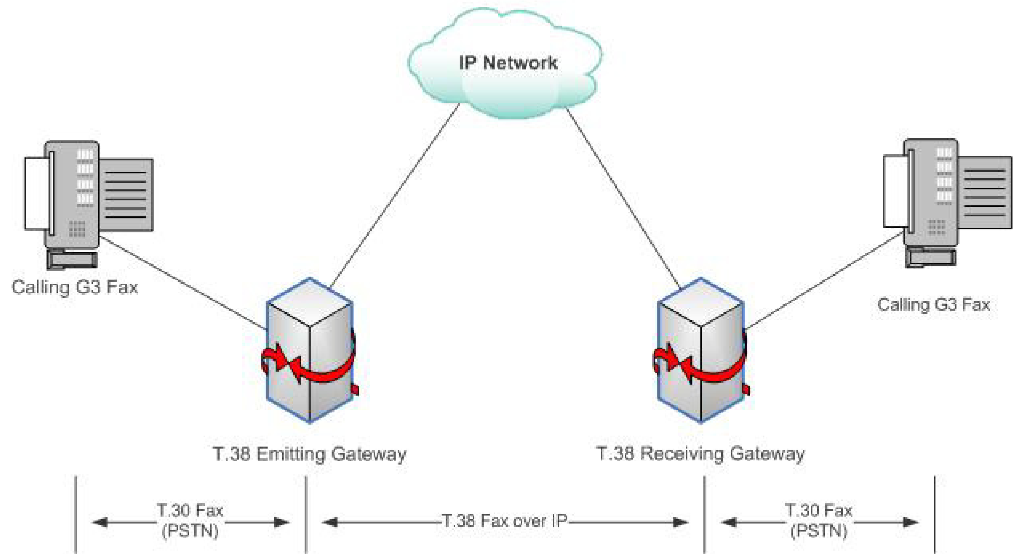


Figure 5-19T.38 Over IP Network Connecting to Fax Machines

5.7.1 Usage

T.38 fax feature is enabled by default on each ShoreTel switch. Only the UDPTL format is supported, with fixed redundancy configurable for all calls. If a secondary fallback extension is necessary, it will be configured as well.

T.38 is implemented through a gateway on the ShoreTel switches. T.38 UDPTL packets parameters are negotiated in the SDP following the offer/answer exchange model via ShoreSIP/SIP between ShoreTel switches, SIP trunk or a SIP based third party devices, such as IP fax extension.

In order to connect a T.38 fax server into the ShoreTel system, one of the two requirement must be met:

1. The T.38 fax server can fall back to G.711 clear channel.
2. If the fax server is T.38 only, all the switches within the system have to be upgraded to one of the supported voice switches, otherwise, fax call from those switches will never be successful. Please refer to the list of supported voice switches listed under the ShoreTel Switches section in the System Specification chapter.

NOTE In order for fax machines to work properly, the administrator should make sure that only fax machines are connected to extensions with one of the following labels:

- Fax machine
- Fax server
- Non-T.38 fax server
- Non-T.38 data terminal

Extensions that are labeled as a fax machine can also be used as a site specific fax redirection machine. It is also necessary to make sure that extensions designated as fax extensions are not fowarded to other phones or trunks using the Anyphone feature. Under these circumstances, fax operation is impacted.

5.7.2 Important Considerations

T.38 support is subject to the following considerations:

- The following ShoreTel Voice Switches do not support T.38. For these and older switches, G711/L16 clear channel is used for fax purposes.
  - — ShoreTel - 8
  - — ShoreTel -12
  - — ShoreTel 120
  - — ShoreTel -T1
  - — ShoreTel -E1
  - — ShoreTel -TW
  - — ShoreTel -24 and ShoreTel 24a
- Fax machine/Fax server behind ShoreTel PBX should disable V34 feature to avoid Fax using G711/Linear clear channel.
- V.34 Faxes are not supported.
- ShoreTel only supports T.38 in udptl form. T.38 calls in RTP or TCP form are not supported.
- ShoreTel does not support either IP media or RFC2833 based fax tone detection (in RFC2833, ShoreTel only supports DTMF, no named telephony events), therefore ShoreTel can not detect Fax tone coming from an SIP end-point with the exception of a SIP connection that is established with a physical port on the ShoreTel switch. In this case, the ShoreTel switch can detect a fax tone from the SIP endpoint and either switch to fax mode or redirect the call.
- ShoreTel depends on fax CNG tone detection or T.38 invite to redirect an incoming fax call. If the fax connection is established with one SIP based endpoint (such as SIP extension or SIP trunk), ShoreTel depends on SIP invite to either establish a fax connection or redirect the call to a pre-configured fax device.
- T.38 support is not supported on SIP-BRI.

5.7.3 Enabling T.38 On a ShoreTel Switch

To enable T.38 support on a ShoreTel switch:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration** > **Call Control** > **Codec Lists**. The Codec Lists page appears as shown ing Figure 5-20.

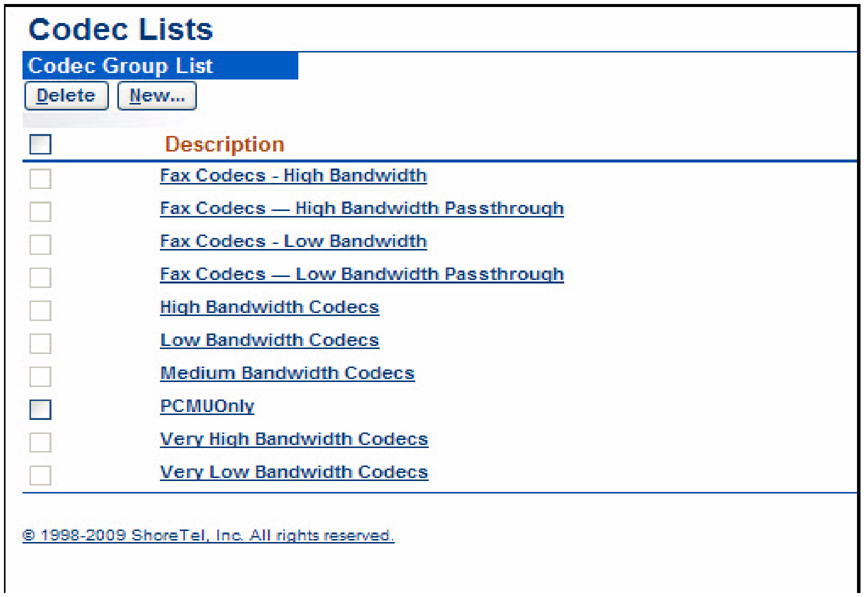


Figure 5-20Codec Lists Page

Step 3 In the Description column, select the fax codec profile for which you want to enable T.38 support or click New to create a new codec profile. The Edit Codec Lists page appears as shown in Figure 5-21.

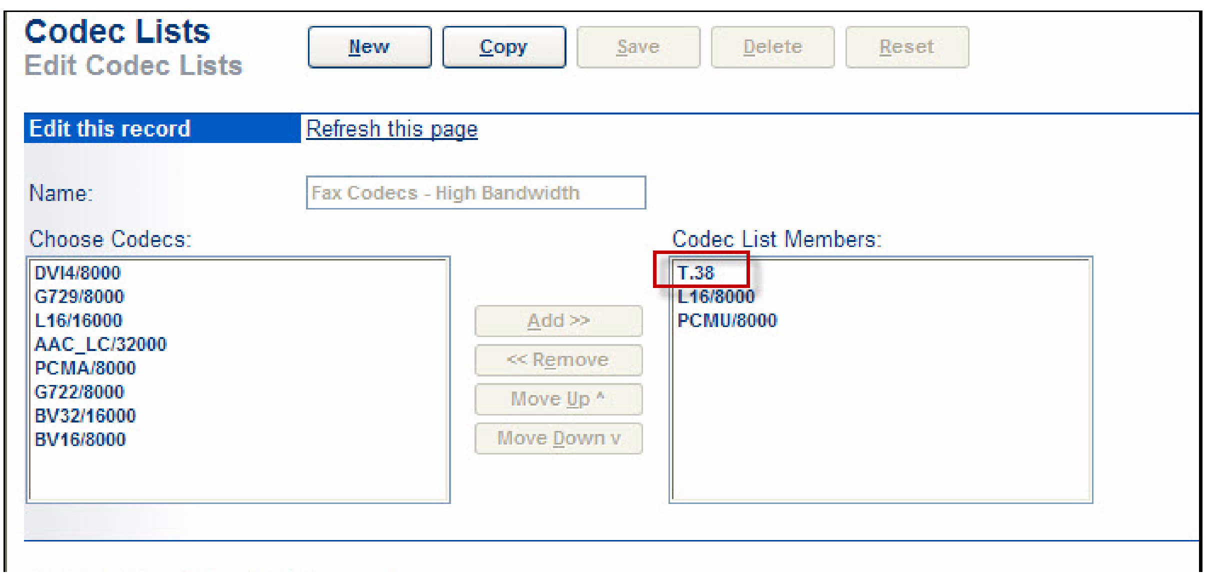


Figure 5-21Edit Codecs List Page

Step 4 Make sure that T.38 is listed in the Codec List Members field and that it is positioned in the order of preference that you want the switch to use it for fax calls.

Step 5 Click **Save**.

Step 6 Click **Administration** > **Sites**. The Sites page appears.

Step 7 Select the site on which you want to enable the T.38 codec. The Edit Site page appears as shown in Figure 5-22

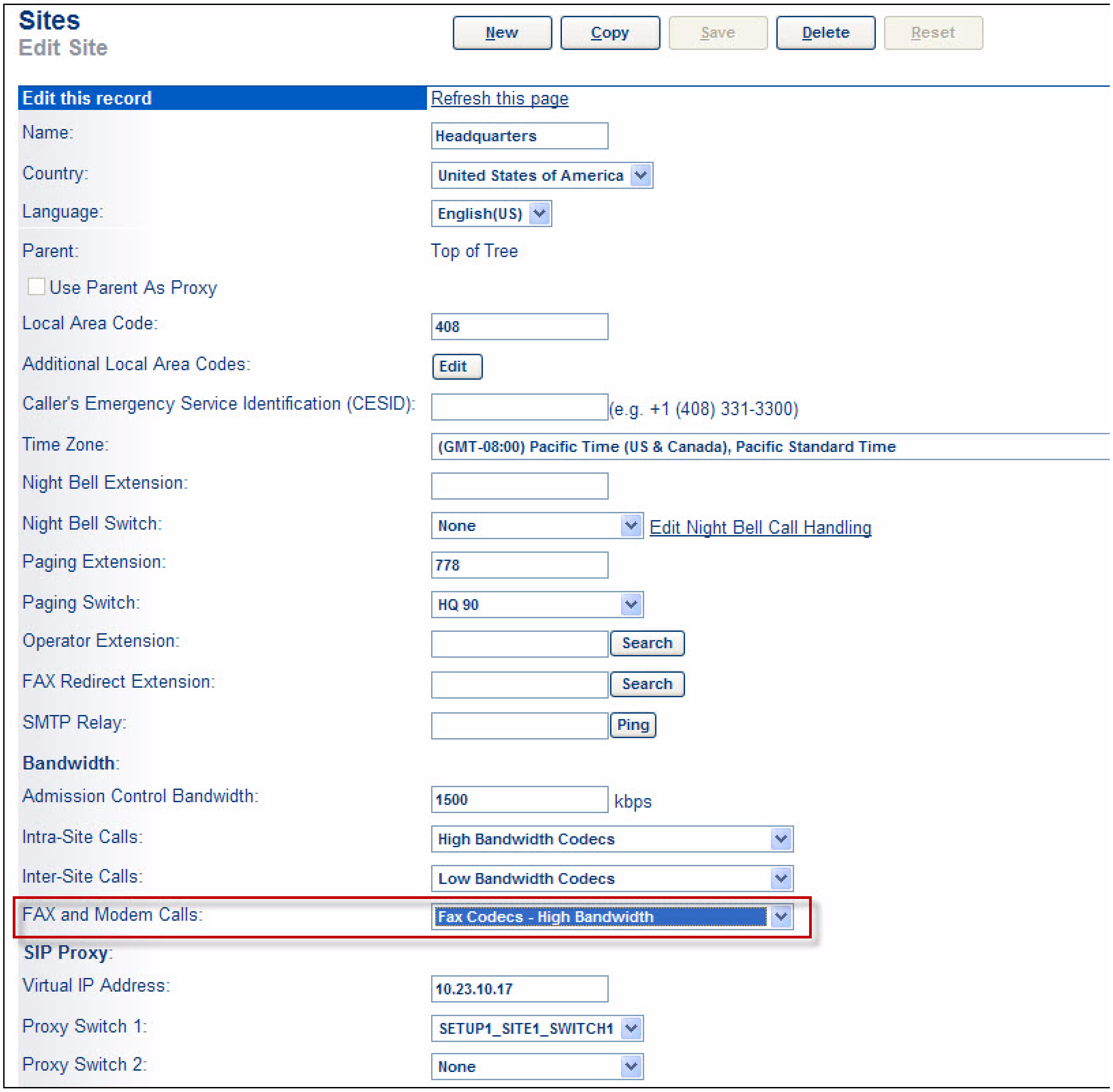


Figure 5-22Edit Site Page

Step 8 In the Fax and Modems Calls field, select a Fax profile in which the T.38 codec is enabled.

Step 9 Click **Save**.

5.7.4 Third-Party T.38 Configuration Support

ShoreTel T.38 implementation also supports GFI Software/Brooktrout SR140. For the configuration procedure, see the following ShoreTel application note:

- ST-10238: How to configure GFI Software/Brooktrout SR140 with the ShoreTel System

Contact ShoreTel Technology Partners Program at http://www.shoretel.com/partners/technology/certified_partners.html for more information on configuring additional ShoreTel supported third-party solutions.

CHAPTER 6

Voicemail Model Switches

This chapter discusses ShoreTel voicemail model switches. The topics include:

6.1 Introduction

Voicemail Model Switches are ShoreTel switches that provide voicemail services and access to auto attendant menus for extensions hosted by the switch. Voicemail Model (V Model) switches provide local access to voicemail while being controlled by a Distributed server at a different location.

Voicemail Model switches store voicemail in Compact Flash (CF) cards. Auto Attendant menus, greetings, and prompts, are stored in permanent flash memory. Voicemail backup and restore routines are available through Director for protecting voice mail on a regular basis. If a switch is disabled, information on the Compact Flash is retained and can be moved to another switch of the same model.

V Model switches are deployed in the same manner as other ShoreTel 1-U Half Width switches and managed similarly to other switches and servers. Director windows configure switch, voicemail, and server settings. Device status is also monitored through Director maintenance windows.

The following ShoreTel switches operate as both a ShoreTel Voice Switch and as a voice mail server:

- ShoreTel Voice Switch 90V
- ShoreTel Voice Switch 90BRIV
- ShoreTel Voice Switch 50V

A ShoreTel site is an organizational concept to which voice switches and users are assigned. ShoreTel Director supports configuration parameters for each site, including time zone, area codes, emergency numbers, voice mail, and local system extensions. Users assigned to a site are associated with the parameter settings for that site. Ports and phones on a switch at a site are available for assignment as the home port of a user assigned to that site.

ShoreTel provides automated attendant and voice mail services through application servers. Although each application server is assigned to a ShoreTel site, an application server can provide services to any ShoreTel user within the ShoreTel network regardless of the site assignments of the user and the server. Therefore, ShoreTel does not require the installation of application servers at each site to provide Auto-Attendant and voice mail services to all system users.

Although users can access voice mail and automated attendants through servers at remote locations, service availability can be subject to time delays and outages inherent in extended networks. Conversely, installing application servers at every site is an inefficient solution for ShoreTel implementations that have several sites configured with a small number of users.

6.2 Functional Description

6.2.1 Physical Description

Voicemail Model switches are similar to other 1-U Half Width Switches, with the addition of permanent flash and Compact Flash memory for providing local access to voicemail, automated scripts, and other services normally provided by Distributed Servers.

6.2.1.1 Capacity

The following are V Model switch capacities:

System

- Maximum Voicemail Enabled Switches in System: 50
- Number of Simultaneous Calls to Voicemail boxes on a V Model switch: 9

ShoreTel 90V

- Analog Telephony: 12 ports
  - 8 ports support trunks,
  - 4 ports configurable as extensions or DID trunks
- IP and SIP resources
  - 90 IP phones, 90 SIP trunks, or 1800 SIP proxies maximum – requires reallocation of telephony resources.
  - 30 IP phones, 30 SIP trunks, or 600 SIP proxies independent of telephony support
- 90 voice mailboxes

ShoreTel 90BRIV

- Analog Telephony: 4 extension ports
- Digital Telephony: 4 BRI ports – 8 channels
  - Each port supports one BRI span that comprises two channels

- IP and SIP resources
  - — 90 IP phones, 90 SIP trunks, or 1800 SIP proxies maximum – requires reallocation of telephony resources.
  - — 30 IP phones, 30 SIP trunks, or 600 SIP proxies independent of telephony support
- 90 voice mailboxes

ShoreTel 50V

- Analog Telephony: 6 ports
  - — 4 ports support trunks,
  - — 2 ports configurable as extensions or DID trunks
- IP and SIP resources
  - — 50 IP phones, 50 SIP trunks, or 1000 SIP proxies maximum – requires reallocation of telephony resources.
  - — 20 IP phones, 20 SIP trunks, or 400 SIP proxies independent of telephony support
- 50 voice mailboxes

6.2.1.2 Differences from other Switches

V Model Switches differ from other 1-U Half Width Switches as follows:

- V Model switches have a slot on the left side of the chassis for accessing the CF card.
- V Model switches provide Voicemail and auto attendant services normally provided by the Main Server or a Distributed Server.
- V Model switch and server functions run on Linux.

  The OS on other ShoreTel switches is VxWorks. Other ShoreTel Servers run under Microsoft Windows.
- V Model switches do not support Simplified Message Desk Interface (SMDI).

6.2.2 Voice Switch Functions

V Model switches provide the same voice switch services as other 1-U Half Width Switches.

V Model switches store voicemail and Auto-Attendant information in 8-bit .wav (µ-law) format received through G.711 and G.729 codecs. The switches can negotiate ADPCM (DVI4/8000) and Linear (L16/8000) codecs. Each switch supports G.729 on trunks and extensions.

Each V Model switch utilize only the codecs residing on that switch. As with other 1-U Half Width Switches, V Model switch codecs cannot serve as a G.729 proxy.

6.2.3 Server Functions

V Model switches support a reduced set of the server features available in ShoreTel Main and Distributed Servers. The following sections describe V Model Switch server features.

6.2.3.1 Voicemail

V Model Switches provide voicemail access to local users under normal operation conditions. Application servers at remote locations can provide services to V Model switch users when switch resource utilization is at capacity.

Switch functions and Server routines run under Linux. Voicemail Model Switches use Qmail, instead of SMTP server used by other Application Servers. V Model Switches do not support SMDI.

Voicemail box capacity varies by switch model:

- SG 90V: 90 mail boxes
- SG 90 BRIV: 90 mail boxes
- SG 50: 50 mail boxes

Voicemail file capacity depends on Compact Flash card size. 1-GByte cards can store up to 1500 minutes of audio data, translating into an average of more than 15 minutes for each SG 90V user.

Voicemail services are provided directly to users from the switch through the IP Phone connected to the V Model switch. When users access voicemail through their computers, the V Model switch sends the file to a Main or Distributed Server, which then transmits the message to the PC.

When the Compact Flash is full, callers attempting to leave voice messages are told the mailbox is full.

6.2.3.2 Auto Attendant Menus

All auto attendant menus defined for the system are loaded onto each V model switch.

6.2.3.3 Recorded Name Storage

When configuring their voicemail boxes, users record their names to audio files for system usage when introducing their mailbox or messages to inbound callers or other users.

V Model switches stores Recorded Name files only for users whose mailbox is hosted on the switch, whereas Main and Distributed Servers retain all Recorded Name files. V Model Switches retrieve Recorded Name files from the Main or Distributed Server when it requires a file that does not reside on the switch.

6.2.3.4 Voicemail Prompts

All voicemail prompts, for languages supported by the system, are stored on the Main Server and each Distributed Server. V Model Switches store prompts for four languages, including the default language of the site to which the switch is assigned.

6.2.4 Connectivity Requirements

Voice mail and auto attendant availability requires connectivity to the boot time server for reading the configuration database on the HQ server. Voicemail and the auto attendant are not active until initial connectivity to the HQ server is established. V Model switches do not require restarting to enable Voicemail and auto attendant if initial connectivity is established after the initial boot.

Voicemail and auto attendant services require that the V model switch has connectivity with a Network Time Protocol (NTP) server. Backup requires FTP server connectivity. As data is backed up, it goes to the Main Server or to any computer with FTP server capabilities that supports RFC 959, the MDTM command, and the SIZE command.

While Distributed servers can manage the V model switch, switch applications depend on access to the database maintained on the Main server. Applications that run on these voice switches include voicemail and Telephone Management Server (TMS).

Personal Communicator connects only to the Main Server or a Distributed Server, even for users whose host port is on a V Model switch.

6.3 Utilities

6.3.1 Accessing Voicemail Model Switch Utilities

ShoreTel switch utilities are accessible through the Maintenance port, an SSH client, or an MS windows program executed from a command prompt on the Main or a Distributed server. The following sections describe utility access methods.

The switch accepts requests from MS Windows CLIs only when they run on the local host, the controlling Distributed server, or the Main ShoreTel server. The switch accepts requests from remote CLIs run from an SSH client.

6.3.1.1 Accessing Utilities from the Serial Port

Switch utilities and the UBOOT command interface are accessible through the maintenance port located on the faceplate. The state of the switch at the time of Maintenance port access determines the available utility.

- During normal switch operation, the maintenance port accesses a specified Linux shell. The default shell is the ShoreTel command line interface (STCLI).
- During a switch boot, the maintenance port accesses UBOOT.

To access ShoreTel utilities through the maintenance port:

Step 1 Connect one end of a serial cable to a computer with a terminal emulator program installed.

Step 2 Connect the male end of the serial cable to the maintenance port on the front panel of the ShoreTel switch.

Step 3 Launch the terminal emulation using the following settings:

- Speed: 19.2 Kbs
- Data bit: 8 bits
- Stop bit: 1
- Parity: No parity
- Flow Control: None

Click **OK**. The ShoreTel command line interface appears.

Step 4 Do one of the following:

- If the interface shows that the switch has a Linux operating system, do as follows
  - — At the command line, type **STCLI**. The STCLI interface appears.

— Type the user ID and password as required. The default values are "admin" and "root" respectively.

- If the interface shows that UBOOT is being used, user ID and password are not required.

Refer to the "STCLI" on page 127 for a description of STCLI. UBOOT is described in the ShoreTel Maintenance Guide.

6.3.1.2 Accessing Utilities from SSH

ShoreTel provides access to several Voicemail Model utilities through a Linux BASH command line. Voicemail Model switches define two accounts:

- **Admin:** The admin account provides access to selected ShoreTel and Linux utilities, including all Voicemail Model command line interfaces. ShoreTel recommends that user log into the Admin account when accessing Linux utilities.

  Logging into the Admin account immediately opens the STCLI interface.

- **Root:** The root account provides access to all ShoreTel and Linux utilities. Access to this account should be restricted because of the potential for creating unintended switch problems.

  Logging into the Root account immediately opens a Linux BASH shell.

Access to the Linux BASH command line through an SSH client. Free SSH clients, such as PuTTY, are available through the internet.

To access a Linux BASH Shell account:

Step 1 Open a SSH client access page. The PuTTY Configuration page appears as shown in Figure 6-1.

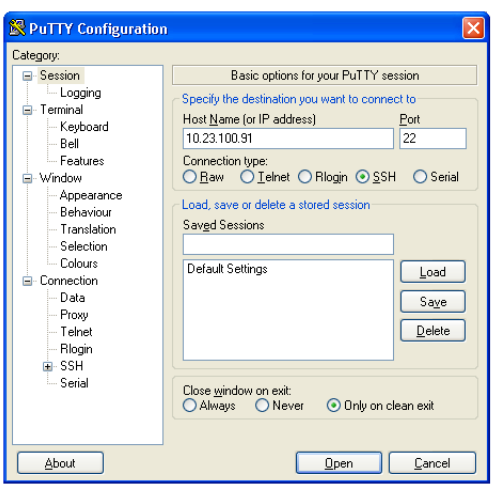


Figure 6-1 PuTTY Configuration Page

Step 2 Open the command prompt window by performing the following:

- Enter the IP address of the switch in the Host Name field
- Enter 22 in the Port field; SSH client communicates on port 22.
- Press the Open button.

Step 3 Enter the username of the desired account on the command line, then press Enter. The Putty command prompt appears as shown in Figure 6-2.

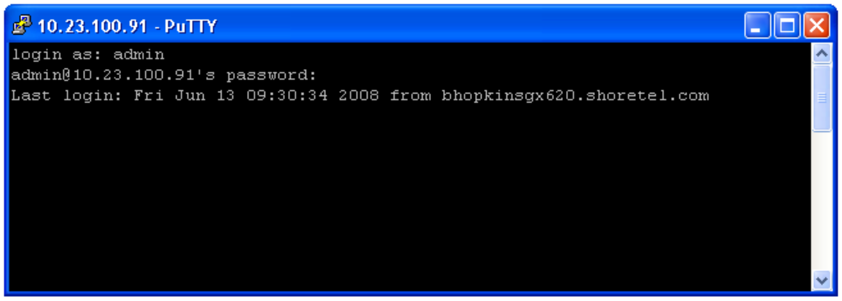


Figure 6-2 PuTTY Command Prompt Window

Where ***admin*** is the entered account.

The command line response depends on the account to which you log into. When logging into admin, the CLI open STCLI. When logging into the Root account, the CLI displays a prompt that displays root as the account.

6.3.2 STCLI

The STCLI is the ShoreTel command line interface (CLI) that allows you to view switch configuration information, manually set the IP address, reboot or shutdown the switch, and archive log files.

When you launch the STCLI as described in "Accessing Utilities from the Serial Port" on page 125 and "Accessing Utilities from SSH" on page 126, the main STCLI menu appears as shown in Figure 6-3.

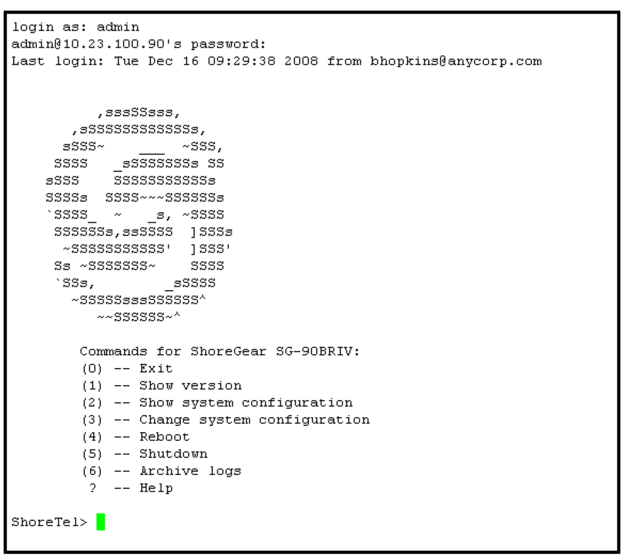


Figure 6-3 STCLI Login and Main Menu

Exiting STCLI returns the user to the Admin account BASH shell. To close the window, type Exit on the Linux command line.

The following describes the **STCLI** commands.

- **Option 0 – Exit**

  This command logs out of STCLI and returns control to the program from where STCLI was entered.

  A user must exit STCLI before starting svccli.

- **Option 1 – Show Version**

  This command displays the system software version running on the V model switch.

- **Option 2 – Show System Configuration**

  This command displays current values for system parameters that are viewable through **STCLI**, a user enters a 2 at the **STCLI** prompt. Figure 6-4 displays an example of the parameters. Option 3 – Change System Configuration provides access to editable parameters.

```
ShoreTel> 2

Current system configuration:
Boot method            = FLASH
Boot command           = jboot;bootm
Boot file              = uImage
DHCP                   = disabled
FTP user name          = anonymous
FTP password           = tsk
Ethernet address       = 00:10:49:08:14:DD
IP address             = 10.23.100.90
IP subnet mask         = 255.255.255.0
Server IP address      = 10.23.100.102
Gateway IP address     = 10.23.100.1
DNS IP address         = 10.23.0.10
Time server IP address = 10.23.0.10
Ethernet link          = auto-negotiate 100 Mb, full-duplex
Domain name            = VMB90BRI

ShoreTel>
```

Figure 6-4 Current System Configuration

- **Option 3 – Change System Configuration**

  The command accesses a list of options for modifying the system configuration. When option 3 is selected, the cursor displays ***ShoreTel Config*** to indicate that subsequent commands my alter the system configuration. Figure 6-5 displays the system configuration menu.

```
ShoreTel> 3

        Change system configuration:
        (0) -- Return to previous menu
        (1) -- Change IP address
        (2) -- Change IP subnet mask
        (3) -- Change gateway IP address
        (4) -- Change server IP address
        (5) -- Change boot method
        (6) -- Enable/disable DHCP
        (D) -- Set/change domain name
        (P) -- Set/change primary DNS IP address
        (S) -- Set/change secondary DNS IP address
        (T) -- Set/change network time server IP address
        (*) -- Display current configuration
         ?  -- Help

ShoreTel Config>
```

Figure 6-5 Configurable System Parameters

The IP addressing mode is selected from this menu. To specify the addressing mode, select 6 from the ShoreTel Config menu. If static IP addressing is selected, all other Option 3 parameters must be configured. The static addressing configuration persists across upgrades.

The configuration file is cleared if the svccli burnflash command is executed.

If DHCP is enabled, the DHCP server must provide the IP address of the network time protocol (NTP) server.

Pressing 0 from the ShoreTel Config prompt returns the system to the main **STCLI** menu. When exiting the **STCLI** main menu, the user is prompted to confirm all configuration changes made in the Option 3 menu.

- **Option 4 – Reboot**

  Option 4 reboots the switch. The switch requests a confirmation of the command before rebooting.

- **Option 5 – Shutdown**

  Option 5 performs a graceful shutdown of the switch. This command is accessible only through the Maintenance port.

  Perform this command before removing power from the switch.

- Option 6 – Archive logs

  Option 6 archives all switch logs and uploads them to the Logs directory in the ftproot of the Headquarters server.

- **Option ? – Help**

  Entering ? lists the main menu items.

6.3.3 Specifying a Static IP Address

The default configuration of new ShoreTel voice switches is to use DHCP to obtain an IP address. To ensure that a DHCP lease is not lost, ShoreTel recommends that you assign a permanent or static IP address to voice switches. You can use DHCP reservations to assign a permanent IP address to a unit or yu can manually assign a static IP address. This section describes how to set a static IP address using the ShoreTel command line interface (STCLI).

To configure a switch with a static IP address using STCLI, do the following:

Step 1 Access the STCLI command line interface, as described in "STCLI" on page 127.

Step 2 Type **3** on the command line to select Change System Configuration. The STCLI window displays the Change System Configuration options.

Step 3 Type **6** on the command line to select Enable/Disable DHCP. The **STCLI** window displays the DHCP options.

Step 4 Type **0** on the command line to select Manual Configuration.

Step 5 Change the network parameters as required to support the fixed address from the Change System Configuration entry line.

Step 6 After completing changes to the configuration, type **exit** to close the STCLI.

Step 7 Reboot the switch.

6.4 V Model Switch Implementation

For a V model switch, Director supports the following tasks through the listed windows:

- Adding a new V model switch to a ShoreTel server, described in "Configuring a Switch" on page 130
- Configuring voice mail, described in "Configuring Voice Mail" on page 132
- Specifying Linux root and administrator passwords, described in "Specifying Root and Administrator Passwords" on page 134
- Specifying maximum size and age of log files stored on the CF card, described in "Log File Size and Age" on page 141
- Configuring backup of voice mail and enabling its automatic activation, described in "Configuring Automatic Backup for a Switch" on page 135.

Monitoring memory usage on the CF card, described in "Monitoring CF Memory Usage" on page 141.

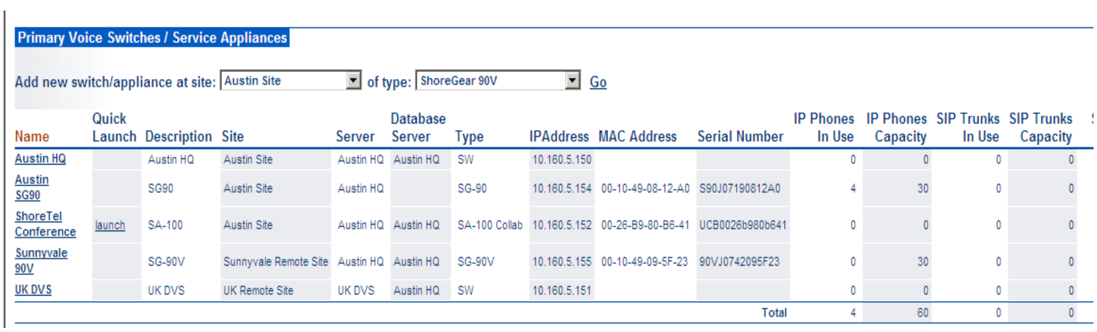
6.4.1 Configuring a Switch

After physically connecting a V Model Switch into the system, the switch is added to the system through Director.

To add and configure a V Model switch to a ShoreTel system:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Platform Hardware > Voice Switches/Service Appliances > Primary**. The Primary Voice Switches/Service Appliances page appears as shown in Figure 6-6.


Primary Voice Switches / Service Appliances

Add new switch/appliance at site: Austin Site of type: ShoreGear 90V Go

Name	Quick Launch	Description	Site	Server	Database Server	Type	IPAddress	MAC Address	Serial Number	IP Phones In Use	IP Phones Capacity	SIP Trunks In Use	SIP Trunks Capacity
Austin HQ		Austin HQ	Austin Site	Austin HQ	Austin HQ	SW	10.160.5.150			0	0	0	0
Austin SG90		SG90	Austin Site	Austin HQ		SG-90	10.160.5.154	00-10-49-08-12-A0	S90J07190812A0	4	30	0	0
ShoreTel Conference	launch	SA-100	Austin Site	Austin HQ	Austin HQ	SA-100 Collab	10.160.5.152	00-26-B9-80-B6-41	UCB0026b980b641	0	0	0	0
Sunnyvale 90V		SG-90V	Sunnyvale Remote Site	Austin HQ	Austin HQ	SG-90V	10.160.5.155	00-10-49-09-5F-23	90VJ0742095F23	0	30	0	0
UK DVS		UK DVS	UK Remote Site	UK DVS	Austin HQ	SW	10.160.5.151			0	0	0	0
									Total	4	60	0	0

Figure 6-6 Primary Voice Switches/ Service Appliances Page

Step 3 In the "Add new switch/appliance at site" field, select the site where you want to add the switch.

Step 4 In the "of type" field, select the model **of the** switch that you want to add to the site.

Step 5 Click **Go**. The Edit Switch page appears as shown in Figure 6-7.

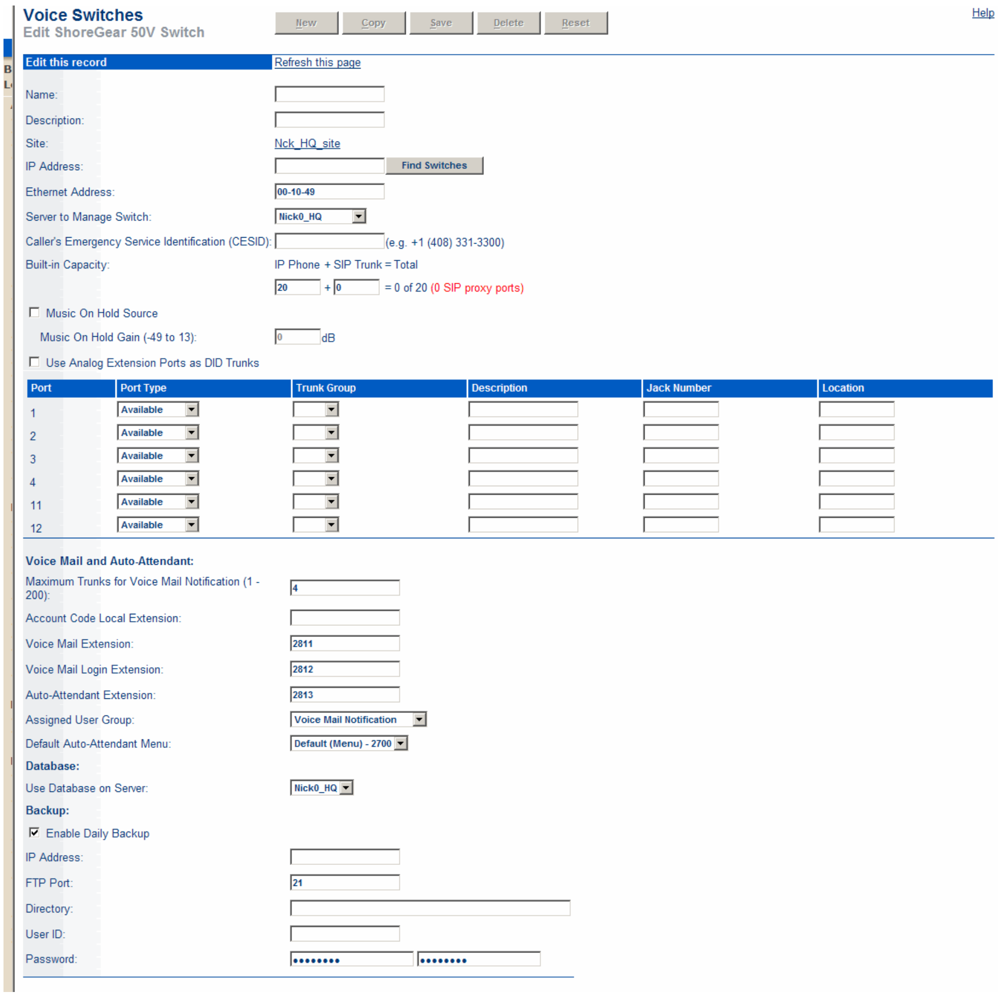


Figure 6-7 Edit Switch Page

Step 6 Complete each field according to the plan for this switch and network.

Refer to "ShoreTel Director Windows for Voice Switches" on page 93 for instructions on configuring a ShoreTel Voice Switch.

The switch configuration window for Voice Model Switches contains voice mail and back-up options in addition to voice switch options available for other switches. Refer to "Configuring Voice Mail" on page 132 for instructions on configuring voice mail. Refer to "Configuring Automatic Backup for a Switch" on page 135 for instructions on configuring system back-up.

6.4.2 Replacing a Switch

When replacing a V Model switch, you can retain the voicemail contents on a CF card if the replacement switch is the same model as the original.

To replace a V Model switch and retain the voicemail on the original switch:

Step 1 Remove the original switch from the ShoreTel network.

Step 2 Remove the plate covering the memory slot on the left side of the original switch.

Step 3 Remove the CF card from the memory slot.

Step 4 Remove the plate covering the memory slot on the left side of the replacement switch.

Step 5 Insert the CF card into the memory slot and replace the memory slot cover.

Step 6 Connect the replacement switch into the network.

Step 7 In ShoreTel Director, open the Switches window by selecting **Administration-> Switches**.

Step 8 Open the Edit ShoreTel Switch page by clicking the name of the replaced switch.

Step 9 Enter the MAC address of the new switch in the Ethernet Address field and press the Save button at the top of the page.

6.4.3 Upgrading a Switch

Upgrading a switch uploads new switch firmware and server software to the device. Switch upgrades are necessary when to maintain compatibility with the remainder of the system when the ShoreTel system is upgraded. V Model switches provide the following upgrade methods:

- Restart and Reboot operations

  Restart and reboot options that also upgrades switch software and firmware when a new version is available in the system. Restart and Reboot are initiated through Director Switch Maintenance pages.

Upgrading the firmware of a V Model switch located at a remote site requires a minimum bandwidth of 384 kbps between the V Model switch and the FTP server providing the firmware upgrade. A remote upgrade may not be possible if the minimum required bandwidth is not available. The firmware upgrade requires 45 minutes at the minimum bandwidth; higher bandwidth reduces the upgrade time.

Existing voice switches cannot be upgraded to become a V model switch.

6.4.4 Configuring Voice Mail

Voicemail is configured through one of the following Director pages:

- **Edit Application Servers**, accessed by selecting Administration > Application Servers > HQ/DVS, then selecting the desired V Model Switch.

  The V model features in the Application Servers window are the same as on other platforms except that on a V model switch:

  — The V model includes the enable and configuration of automatic daily backup.

  — Simplified Message Desk Interface (SMDI) is disabled.

  Figure 6-8 displays the Application Servers window for the V Model switch named HQVMB-50

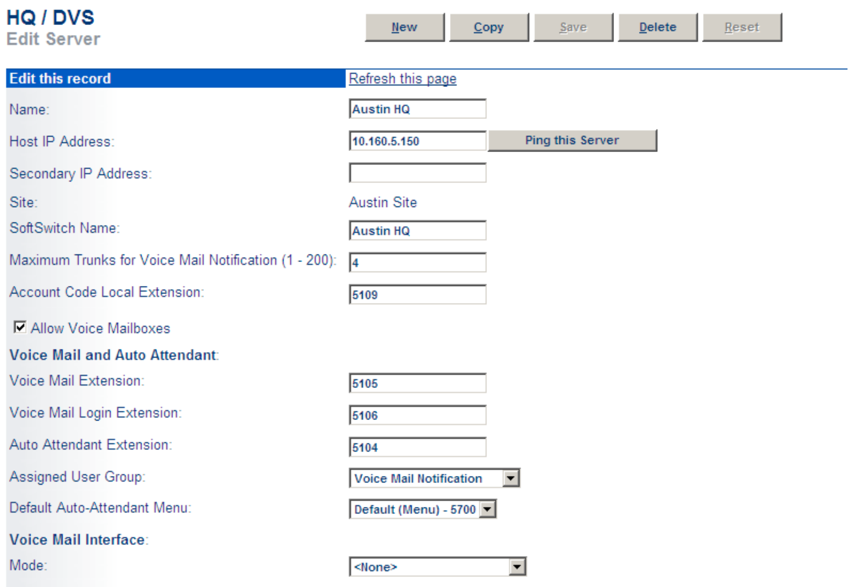


Figure 6-8 Configuring Voice Mail through Application Servers Window

- **Edit Switches**, accessed by selecting Administration > Application Servers > HQ/ DVS, then selecting the desired V Model switch.

  Figure 6-9 displays the Voicemail parameters on the **Switches** window for the V model switch named HQVMB-50.

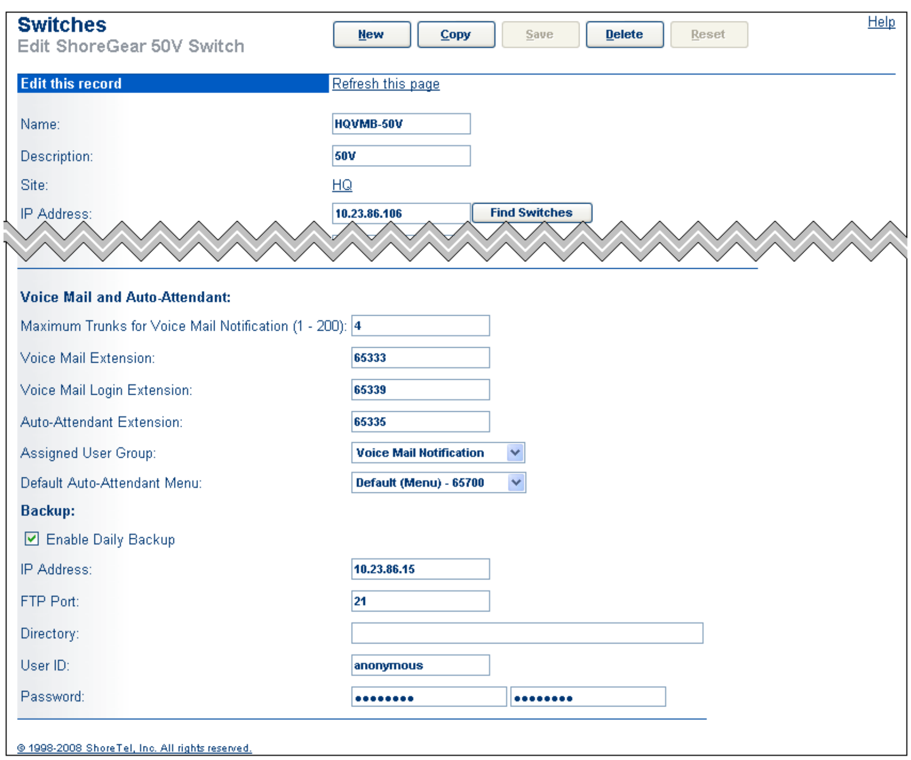


Figure 6-9 Configuring Voice Mail through the Switches Window

Director paths that open voice mail on the Main and Distributed Servers also access pages that configure V model switches. For details on voice mail configuration, see Chapter 6: Voicemail Model Switches, starting on page 121.

6.4.5 Specifying Root and Administrator Passwords

V Model switches provide access to command line interfaces (CLIs) for diagnostics and advanced configuration tasks. Other than specifying a fixed IP Address, CLI access is not required for typical switch operation and maintenance.

ShoreTel provides two default accounts for accessing V Model Switch CLIs:

- **Admin:** This account is used for configuration tasks that require CLI access.
- **Root:** The root account accesses all internal Linux commands.

IMPORTANT **Important:** ShoreTel recommends using the Root command only under direct supervision of ShoreTel Technical Support personnel. The root admin does not restrict command scenarios that can render the switch unusable.

You can use ShoreTel Director to change the passwords for logging into these accounts. To change the passwords required to log into the Voicemail Switch CLI accounts:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > System Parameters > Other**. The Edit Other Parameters page appears as shown in Figure 6-10.

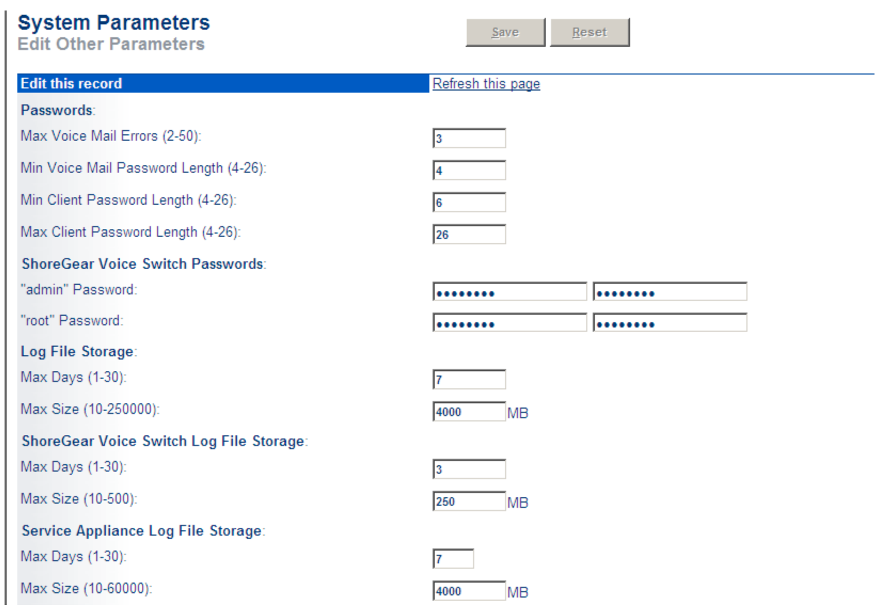


Figure 6-10Edit Other Parameters Page

Step 3 In the first “admin” Password field, enter the password that you want to use for the “admin” account.

The password must have four ASCII characters minimum and up to 26 ASCII characters maximum.

Step 4 In the second "admin" Password field, enter the same password that you entered into the first field.

Step 5 In the first "root" Password field, enter the password that you want to use for the "admin" account.

The password must have four ASCII characters minimum and up to 26 ASCII characters maximum.

Step 6 In the second "root" Password field, enter the same password that you entered into the first field.

Step 7 Click **Save**.

6.4.6 Configuring Automatic Backup for a Switch

Due to the CF card capacity, daily backup of voice mail and Auto-Attendant data is advised. When automatic backup is enabled in the ShoreTel Director, it begins immediately after the server completes its daily house-keeping operation. (The house-keeping utilities are built-in and remove log files and old voice mails based on the configuration data in the Director.) In addition:

Automatic backup stores voicemail, auto attendant data, and switch log files to an FTP server. After completion of the daily file-system cleanup tasks, the switch begins automatic backup. A timestamp is appended to the name of files copied to the target server.

Automatic backup provides a source for the most recent day's voice mail and other data in the event of a system failure. It is not intended to be an archive of voice messages or a source for retrieving deleted voice mail.

NOTE V model switches rely on the ftpsync facility to synchronize its local (or source) directory to the back-up server (or target) directory. Therefore, the server must support ftpsync as described in RFC 959. The server must also support the MDTM and SIZE commands. FTP servers in Win2003 and WIN XP meet these requirements.

To configure automatic backup for a switch, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Platform Hardware > Primary**. The Primary Voice Switch/Service Appliances page appears.

Step 3 Select the switch that you want to configure for automatic backup. The Edit page for that switch appears.

Step 4 Scroll down to the Backup section and do the following:

Step a Check the **Enable Daily Backup** checkbox.

Step b In the IP address field, enter IP address of the FTP server to which you want to back the switch files up.

Step c In the FTP Port field, enter the port number that the switch is to use to communicate with the recipient FTP server.

Step d In the Directory field, enter the path to the file on the FTP server to which you want to back the switch files up.

Step e In the User ID field, enter the user name that the switch is to use to access the FTP server files for backup.

Step f In the first Password field, enter the password that the switch is to use to access the FTP server files for backup.

Step g In the second Password field, enter the same password that the switch is to use to access the FTP server files for backup.

Step h Click **Save**.

6.4.7 Configuring a Target Server for Backup

Backing up V model switch voice mail requires a target FTP server. A ShoreTel Main Server or a third-party server can function as the recipient of V Model Switch backup files.

This section describes an FTP server configuration, using a Windows server as the recipient. (Detailed instructions on configuring Windows servers are available in the article *How To Set Up Isolated FTP Site* at http://support.microsoft.com/kb/555018.) The section first lists the configuration data entered into Director. The subsequent list shows the information entered at the server, including configuration data from the V model switch.

On the server, perform the following:

Step 1 Using the Win2003 Computer Management dialog, add a new FTP site

In Figure 6-11, an FTP site named "ShoreTelBackup" is added.

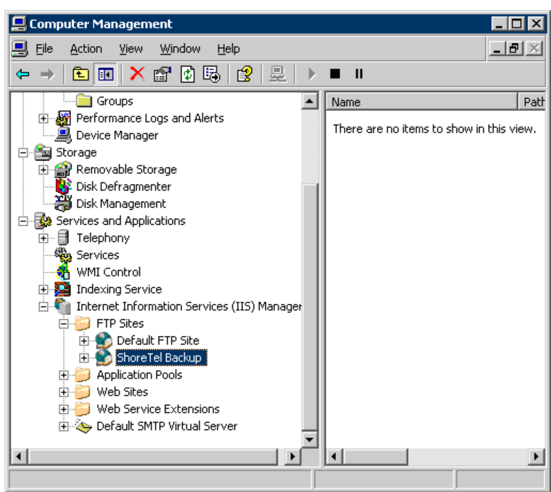


Figure 6-11The New Target Server is Identified

Step 2 For the new server, specify the IP address and port number

In Figure 6-12, the IP address is 10.1.1.42. and port is 5555.

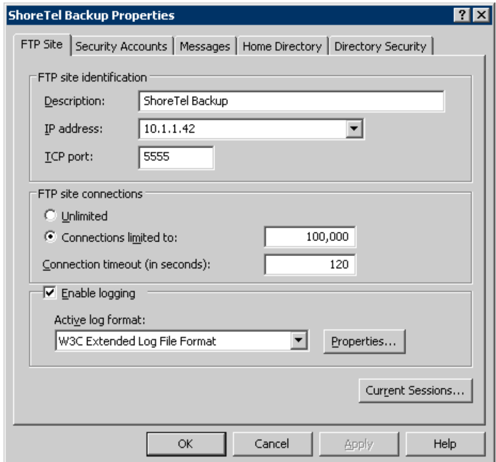


Figure 6-12Specify IP Address and Port Number for "ShoreTelBackup

Step 3 In the ShoreTel Backup Properties page, specify the local path

In Figure 6-13, the local path is C:\ShoreTelBackup.

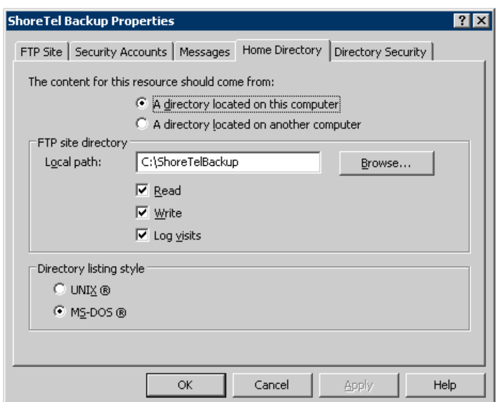


Figure 6-13Specifying the Local Path for the Backed-up Data

Step 4 In the FTP Site Creation Wizard, specify that the user is isolated.

Figure 6-14 displays a properly configured Site Creation Wizard.

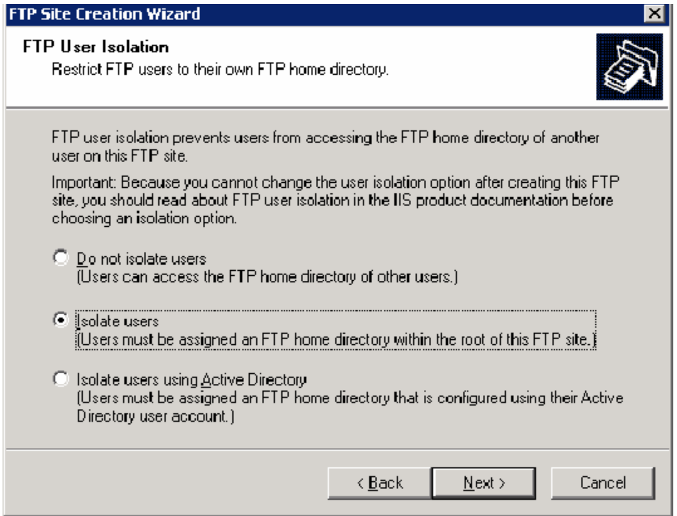


Figure 6-14Isolating the User (Directory)

Step 5 Enter a name and description of the FTP server in the General page, as shown in Figure 6-15.

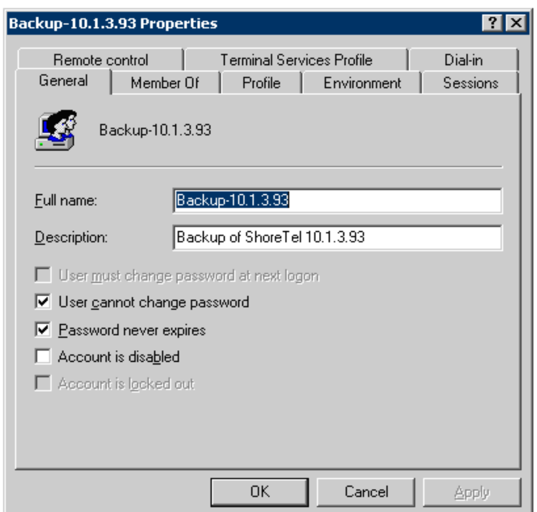


Figure 6-15Specifying the and Password Details

Step 6 Enable the following options:

- User cannot change password
- Password never expires.

Step 7 Verify the resultant path of the configured server, as shown in Figure 6-16.

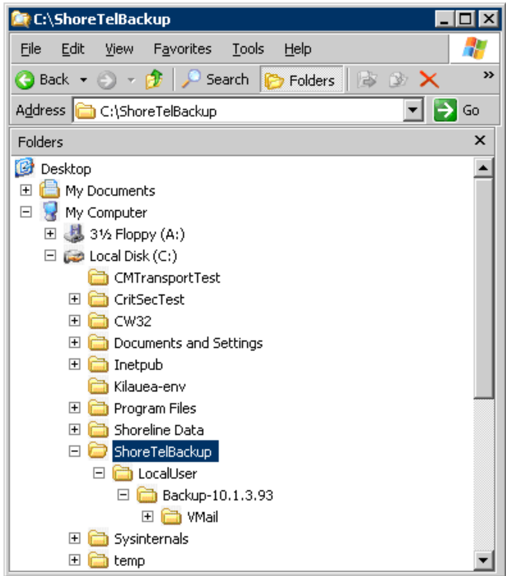


Figure 6-16After Configuring the FTP Server is Finished

6.5 Booting and Restarting

6.5.1 Description

Rebooting and restarting V model switches have different scopes.

- Rebooting a V model switch also reboots the Linux kernel and everything that a kernel reboot entails.

  A reboot takes much longer than a restart.

- Restarting a V model switch only the reboots the ShoreTel application layer.

  On ShoreTel switches running on VxWorks, rebooting and restarting are identical.

Under certain conditions, initiating a restart will reboot the switch. One example is when a switch upgrade is available.

When a ShoreTel voice switch boots, it requires an IP address to connect to the network and an application program. ShoreTel voice switches are set to use a DHCP server for an IP address and to retrieve the application from the switch's flash memory.

If a DHCP request receives no response, the switch tries a BootP request as a backup. ShoreTel recommends using static IP parameters configured via the serial port, as this is much more reliable. When using DHCP, ShoreTel recommends using DHCP reservations for each switch to ensure that DHCP leases are not lost.

If a DHCP server is not available, you can set the IP address manually from the switch's maintenance port from STCLI.

If the switch fails to load the application from flash and does not have the IP address of the ShoreTel server, you can set the IP address and boot parameters by connecting to the maintenance port and using the configuration menu. The configuration menu allows you to set the IP address of the switch and enter the ShoreTel server (boot host) IP address.

A V model switch can be brought up by a regular (flash memory-sourced) boot or by a software upgrade boot.

6.5.2 Specifying a Time Source

ShoreTel servers maintain an internal time of day clock, which is initialized by input received at boot time from an NTP server. ShoreTel servers use the time of day clock to mark voicemail and track other transactions.

V model switches do not begin server operations until after they receive an initial time of day input. If the time is not available from a designated source at boot time, the V Model switch supports all switch operations and will periodically poll for the time of day setting. After receiving the a time of day setting, the V model switch begins server operations.

The NTP server can be specified through DHCP or as a static address. If no address is specified, the V Model switch polls NTP servers at addresses specified by an internal configuration list. The internal configuration list includes the HQ Server and Internet based NTP servers.

- If an IP address is listed that does not point at an NTP server, the V Model switch will not begin server processes until the address is corrected.
- If the IP address points at a server that is not available, the V Model switch periodically polls the IP address for the NTP server. When the server becomes available, the V Model switch begins performing server operations after it polls the server and receives the time of day setting.

  After the server becomes available, rebooting the V model switch may be faster than waiting for it to poll the NTP server.

6.5.3 Reboot Methods

6.5.3.1 Flash Boot

The standard method for booting a ShoreTel voice switch is to boot from the switch's flash memory. When a ShoreTel switch is first powered on, it reads the boot parameters stored on the non volatile memory, which instructs the switch to load software from flash memory. When the software starts, it loads its configuration, which is also stored in flash memory.

6.5.3.2 Default Button

The Default Button is the small “paperclip” button on the left side of the switch. Pressing this button replaces the two configuration files with their default variants. The Compact Flash is not affected.

Pressing this button and holding for 10 seconds, in addition to replacing the configuration files, removes all files from the Compact Flash.

6.5.3.3 FTP Boot

Booting from FTP is available when you cannot boot the switch from internal memory. When booting a switch from FTP, the operating system and software are loaded from the FTP site identified in the boot parameters. The loaded files define a default configuration.

Voicemail services on the switch are disabled after booting from FTP and are restarted only by booting from Flash. After an FTP boot, the switch can perform telephony functions as those available through other ShoreTel switches. V model switches started with an FTP boot can operate only as a voice switch – (controlling phones, trunks, and call routing.

FTP boot is typically used for troubleshooting and also supports maintenance tasks and the backup and restore facilities.FTP boot supports certain maintenance functions, such as an emergency boot if the flash becomes damaged.

6.6 Monitoring a Voicemail Model Switch

6.6.1 Monitoring Switch Functions

Quick Look exists in ShoreTel Director to provide an overview of the ShoreTel system status. It includes information about each site and the corresponding switches, ports, servers, and service. This is the first place to look to determine the status of the system.

6.6.2 Log File Size and Age

The Other Parameters window under Administration > System Parameters contains fields that show the maximum size and age of log files. These values must be set on a server, not in the V-model Director's switch area.

Due to the constraints on CF memory space, log files should not be allowed to consume too much memory. Thereafter, after the V-model switch has been added, change the values for the maximum size and age of log files from a server.

6.6.2.1 Event Logs

The ShoreTel system uses the Windows Event Log, viewed using the Event Viewer, to report information and errors that are of interest to system administrators. You can use the event logs in conjunction with Quick Look to determine the overall health of the system. You can also use the event log to gather information about an event that is no longer a problem. For example, the event log may provide information about an overnight T1 outage that was corrected is no longer evident.

Each system task reports when it starts and stops. These messages can be helpful in determining whether the system has started correctly. Events, such as switches losing connection to the server or rebooting, are also reported.

6.6.2.2 System Logs

A ShoreTel system stores engineering-level log files that record transaction information about every event in the ShoreTel system. The logs help ShoreTel with debugging problems that can arise during system operation. In most cases, these logs are difficult to interpret and require help from ShoreTel Customer Support to understand.

6.6.3 Monitoring CF Memory Usage

A V model switch provides CF usage data to the ShoreTel Director and a CLI command. If the CF card becomes full, it cannot accept new voice mail. This section describes four Director windows that provide different views of memory usage on a CF card. Certain Director windows variously show the CF's used space, free space, percent of usage, and color-coded alerts. Also, the administrator can create an event filter to send notification of high memory usage.

Typical CF capacities are 1 Gbyte, 2 Gbytes, and 4 Gbytes. The proper planning of voice mail usage and aging can help prevent excessive buildup of voice mail on the CF card.

6.6.3.1 Using Quick Look

The Quick Look page in ShoreTel Director lets you see how much of the CF card is currently being used. To view CF card usage in the Quick Look page, do the following:

Step 1 Launch ShoreTel Director. The Quick Look page appears as in Figure 6-17.

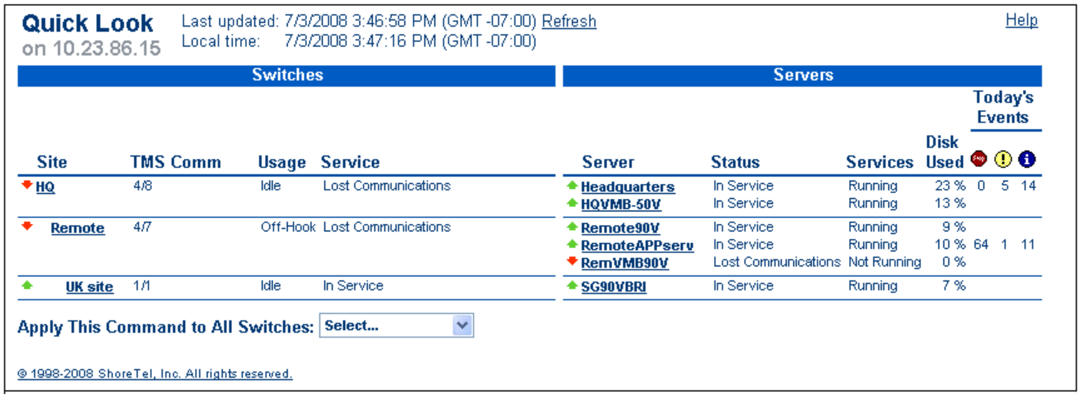


Figure 6-17Percent of Disk Usage in the Quick Look Window

Step 2 Locate the voice switch for which you want to monitor the CF card usage.

Step 3 Check the value in the Disk used column to see how much of the CF card is currently being used.

In Quick Look, the colored alerts are yellow (1–10 messages left) and red (mailbox is full).

6.6.3.2 Using the Maintenance Page for the Switch

The maintenance page for the switch lets you view information about usage by switch users. To view the information on the maintenance page:

Step 1 Launch ShoreTel Director. The Quick Look page appears.

Step 2 Select the switch for which you want to view information. The Maintenance page for the switch appears as shown in Figure 6-18.

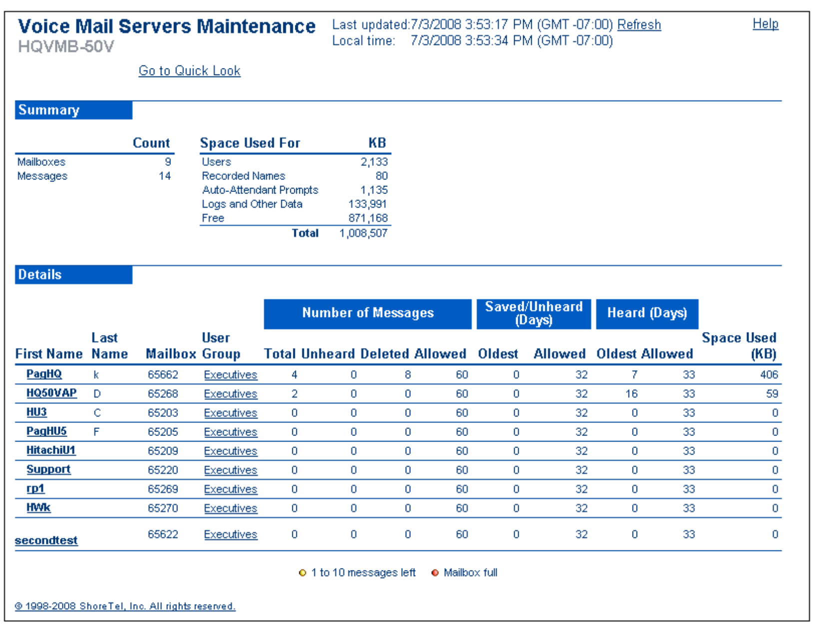
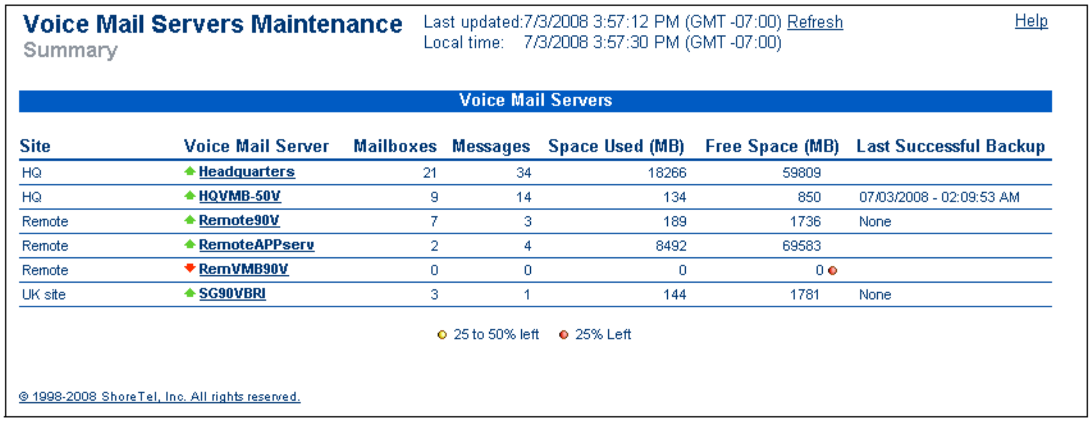


Figure 6-18Specific Voice Mail Availability through the Quick Look Window

In Quick Look, an administrator can select a particular V model switch and see the configuration and usage numbers for individual voice mailboxes (). In other windows, these colors can mean other thresholds.

For the amounts of free and used disk space in Mbytes and any alerts, the administrator can go to the Voice Mail Servers Maintenance - Summary window. This window (Figure 6-19) shows the amount of free and used space in Mbytes for each voice mail server. For memory usage warnings, an alarm indicator appears in the Free Space (MB) column next a specific switch's value. In Figure 6-19, the small circle next to the HQVMB-50V Free Space value is a yellow alert. The legend at the bottom of the screen shows that yellow means CF free space is at 25-50% of capacity. Red means free space is at or below 25% of capacity.


Voice Mail Servers Maintenance
Summary
Last updated:7/3/2008 3:57:12 PM (GMT -07:00) Refresh
Local time: 7/3/2008 3:57:30 PM (GMT -07:00)
Help

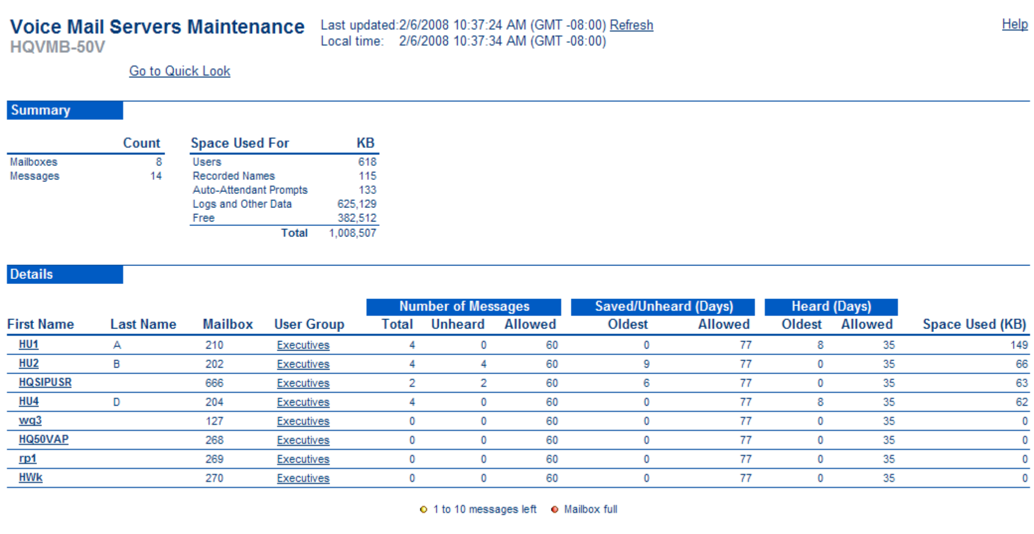
Voice Mail Servers

Site	Voice Mail Server	Mailboxes	Messages	Space Used (MB)	Free Space (MB)	Last Successful Backup
HQ	Headquarters	21	34	18266	59809	
HQ	HQVMB-50V	9	14	134	850	07/03/2008 - 02:09:53 AM
Remote	Remote90V	7	3	189	1736	None
Remote	RemoteAPPserv	2	4	8492	69583	
Remote	RemVMB90V	0	0	0	0	
UK site	SG90VBRI	3	1	144	1781	None

25 to 50% left 25% Left

Figure 6-19Disk Usage for Each Server in Voice Mail Servers Maintenance

The next degree of more granularity is information for each user. In Figure 6-20, the screen for voice mail server HQVMB-50V shows the current number of Kbytes of voice mail for each user and the configuration details for each user. The alerts in this window are yellow for 1–10 messages left and red for mailbox full.


Voice Mail Servers Maintenance
HQVMB-50V
Last updated:2/6/2008 10:37:24 AM (GMT -08:00) Refresh
Local time: 2/6/2008 10:37:34 AM (GMT -08:00)
Help
Go to Quick Look

Summary

	Count
Mailboxes	8
Messages	14

Space Used For	KB
Users	618
Recorded Names	115
Auto-Attendant Prompts	133
Logs and Other Data	625,129
Free	382,512
Total	1,008,507

Details

				Number of Messages			Saved/Unheard (Days)		Heard (Days)		
First Name	Last Name	Mailbox	User Group	Total	Unheard	Allowed	Oldest	Allowed	Oldest	Allowed	Space Used (KB)
HU1	A	210	Executives	4	0	60	0	77	8	35	149
HU2	B	202	Executives	4	4	60	9	77	0	35	66
HQSIPUSR		666	Executives	2	2	60	6	77	0	35	63
HU4	D	204	Executives	4	0	60	0	77	8	35	62
wq3		127	Executives	0	0	60	0	77	0	35	0
HQ50VAP		268	Executives	0	0	60	0	77	0	35	0
rp1		269	Executives	0	0	60	0	77	0	35	0
HWk		270	Executives	0	0	60	0	77	0	35	0

1 to 10 messages left Mailbox full

Figure 6-20CF Card Usage per User on HQVMB-50V

ShoreTel Director displays the CF memory usage as a disk usage value.

6.7 Creating a System Extension for SA-100 Conferences

The ShoreTel system automatically creates a system-wide conference number that it assigns an internal ShoreTel extension. This is a number that all ShoreTel users can use to initiate or join a ShoreTel conference. You can change extension manually and even use an external phone number for the conference extension.

Each SA-100 that you install is also automatically assigned a ShoreTel extension. This number can be used by local users to access the unit to initiate conference calls. You can also change the extension number assigned to the unit.

This section describes how to change the extension and assign an external number.

Requirement

- A valid external phone number properly mapped in ShoreWare Director

To change the internal extension and assign an external number, do the following:

Step 1 Launch ShoreWare Director.

Step 2 Click **Administration > System Parameters >System Extensions**. The Systems Parameters page as shown in Figure 6-21 appears.

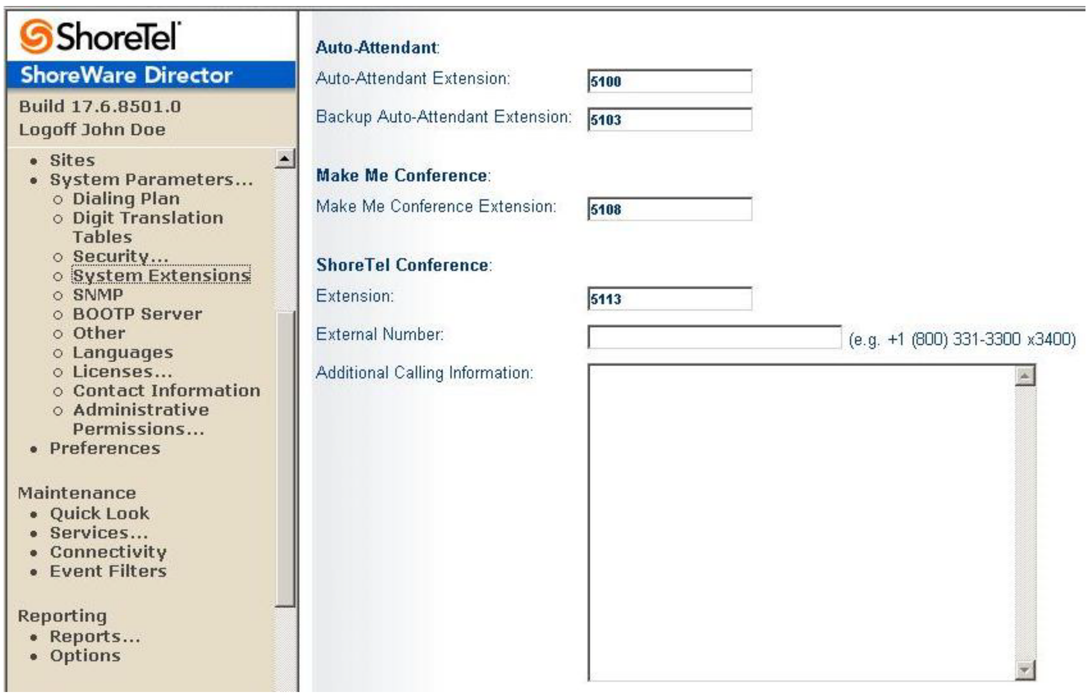


Figure 6-21Systems Parameters Page

Step 3 In the Extension field, enter the available extension number that you want to use for ShoreTel Conference to change the internal extension assigned to ShoreTel Conference.

Step 4 In the External Number field, enter the valid phone number that you want to use for ShoreTel Conference.

NOTE Make sure that the number that you use is properly mapped to a telephony trunk. Refer to the ***ShoreTel 12: System Administration Guide*** for information about setting up trunks.

Step 5 In the Additional Calling Information field, enter any additional information that you want to use with the extension. The text will be included in all conference invites including Outlook invite as well as in the dial in window in the ShoreTel Viewer.

6.8 Configuring the SA-100 for Conferencing and IM

This section describes how to set up the ShoreTel system to use the SA-100 to provide conferencing and IM services. For more information about setting up and configuring the SA-100, refer to the *ShoreTel Service Appliance 100 Planning, Installation, and Administration Guid*e.

NOTE If after you install and configure the SA-100 you use a Windows server machine to access the conferencing Web page, make sure that your browser allows JavaScript to download. JavaScript is required to render the conference Web page.

6.8.1 Creating SA-100 Users

Anyone can participate in or initate ad hoc SA-100-managed conference calls. However, the SA-100 users referred to here are ShoreTel system users who are granted accounts that allow them to schedule and host conferences on SA-100s. To configure these SA-100 users, do the following:

Step 1 Launch ShoreWare Director.

Step 2 Click **Administration > Users > Individual Users**. The Individual Users page appears.

Step 3 Select the user for whom you want to enable the service. The Users page containing the profile of the user you select appears.

Step 4 Scroll down to Conference Setting section near the bottom of the page.

Step 5 In the Appliance field, select the SA-100 that you want the user to use for conference calls.

Step 6 Click **Save**.

Note If you are not running proxy services on your network, we recommend that you disable the Microsoft Internet Explorer Internet Option setting “Automatically detect settings” on the machines of conference users. This setting can cause the Web Presenter on user machines using a WiFi connection to take up to 30 seconds to load during conferences.

6.8.2 Migrating IM and Presence Users to the SA-100 Server

To migrate IM and Presence users from a legacy ShoreTel IM Server to the SA-100, do the following:

Step 1 Launch ShoreWare Director.

Step 2 Click **Administration > Users > Individual Users**. The Individual Users page appears.

Step 3 Select the user that you want to move to the SA-100. The Users page appears.

Step 4 Locate the Instant Messaging Settings section and in the Server/Appliance field, select the SA-100 unit that you want the user to use for IM and Presence.

Step 5 Click **Save**.

Step 6 Notify the end-user to uninstall, then install Outlook calendar integration in ShoreTel Communicator for Windows if they are using Outlook calendar-integration. Refer to *ShoreTel 12:1 Communicator for Windows User Guide* for instruction on how to configure Outlook calendar integration.

6.8.3 Configuring the Conference Web Page

A ShoreWare Director user with administration privilege can customize the Web page that the current SA-100 generates for conferences. The administrator can change the logo that appears, place a link behind the logo that redirects the user to another site, and change the name that appears on the page. To customize the conference Web page for the SA-100, do the following:

Step 1 Log in to ShoreWare Director as a user with administration privilege.

NOTE The default user ID and password are not supported for this application.

Step 2 Click **Administration > Platform Hardware > Voice Switches/Service Appliances > Primary**. The Primary Voice Switches/ Service Appliance page appears as shown in Figure 6-22.

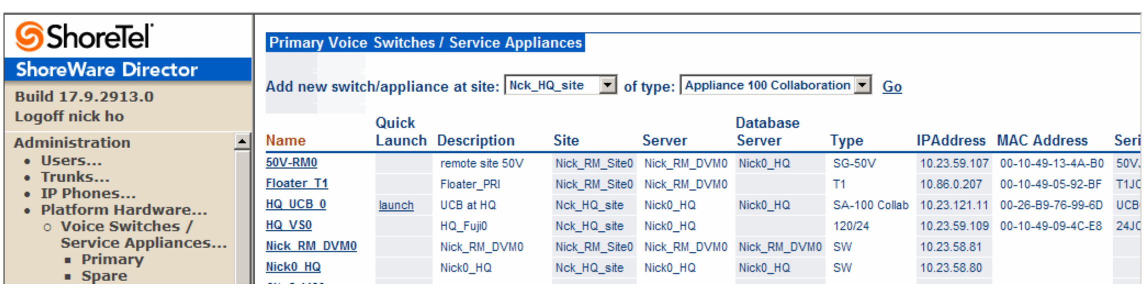


Figure 6-22Primary Voice Switches/Service Appliance Page for SA-100

Step 3 Locate the SA-100 for which you want to modify the conference Web page and click **Launch** in the Quick Launch column. The Service Appliance Conference Administration page appears as shown in Figure 6-23.

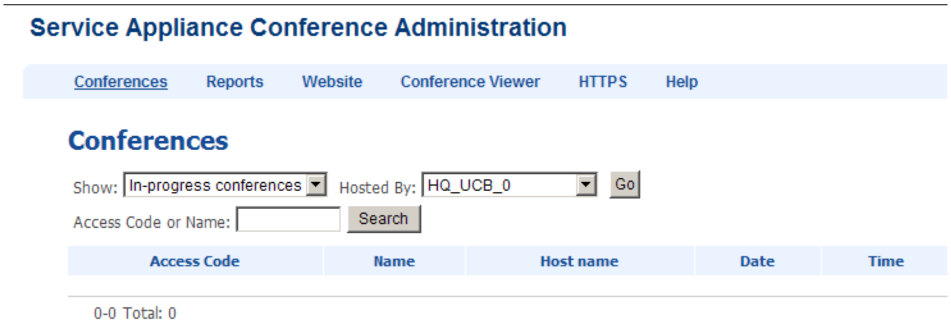


Figure 6-23Service Appliance Conference Administration Page

Step 4 Click **Website**. The Website page appears as shown in Figure 6-24.

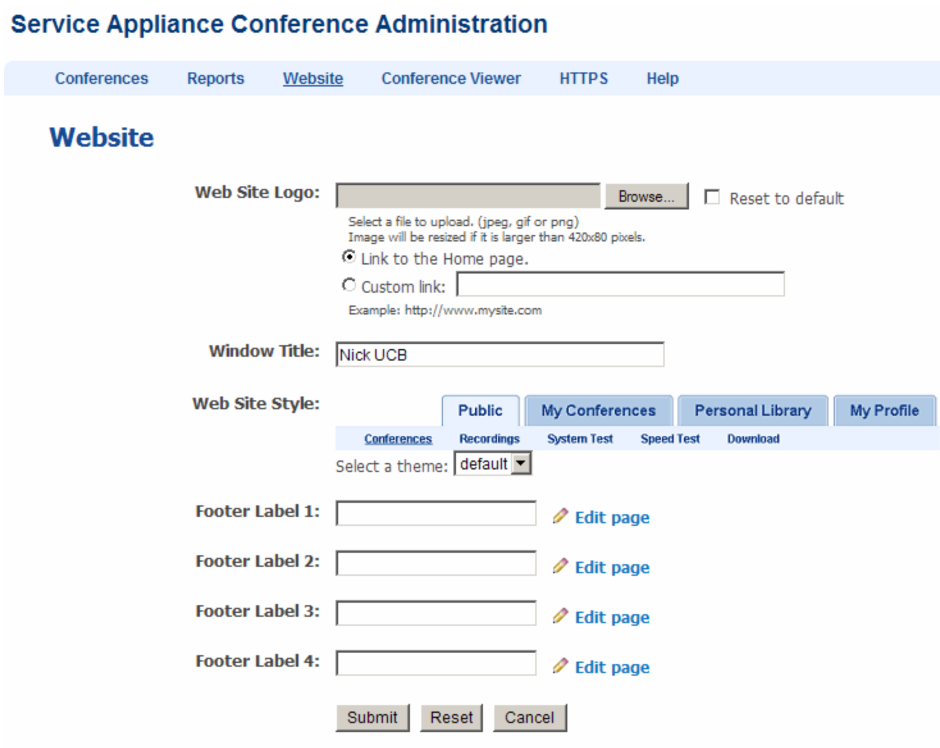


Figure 6-24Website Page

Step 5 In the Web Site Logo field, enter the path to the logo that you want to appear on Web conference pages.

Step 6 To specify the link that you want to associate with the Web site logo:

- Select the **Link to the Home page** radio button to connect users to the conference home page when they click on the logo.
- Select the **Custom link** radio button and enter the path to the destination that you want users to go to when they click on the logo.

Step 7 In the Window Title field, enter the name that you want to use as the identifier for conferences managed by this SA-100. The name will appear on the browser tab of active conferences. The default name is ShoreTel Conference.

Step 8 To create footers that appears at the bottom of the conference Web page, do the following:

Step a In a Footer Label field, enter the name that you want to appear as a footer at the bottom of conference Web pages.

Step b Click **Submit** to create the footer. The page refreshes.

Step c Click **Edit page** next to the footer label for which you want to provide text. The "Edit *footer name*" page appears.

Step d Enter the text that you want to appear when a user clicks the footer. You can use an HTML text to populate the field.

Step e Click **Submit**. The page refreshes.

The footer will appear on the bar at the bottom of the conference Web page. User access the message by clicking the footer label.

NOTE To remove a footer, erase the footer label and click Submit.

Step 9 Click **Submit** to save your changes.

NOTE The remaining fields are not used.

6.8.4 Configuring the Parameters for the Client Interface

You can set interface parameters that affect conference participants. You can insert a logo on the viewer Web page, activate a tone that signals a user joining a conference, and you can set up a link that directs participants to a Web page at the completion of their conference. To configure these parameters for the client interface, do the following:

Step 1 Log in to ShoreWare Director as a user with administration privilege.

NOTE The default user ID and password are not supported for this application.

Step 2 Click **Administration > Platform Hardware > Voice Switches/Service Appliances > Primary.** The Primary Voice Switches/ Service Appliance page appears.

Step 3 Locate the SA-100 for which you want to modify the conference interface and in the Quick Launch column click **Launch**. The Service Appliance Conference Administration page appears as shown in Figure 6-23 on page 147.

Step 4 Click **Conference Viewer**. The Conference Viewer page appears as shown in Figure 6-25.

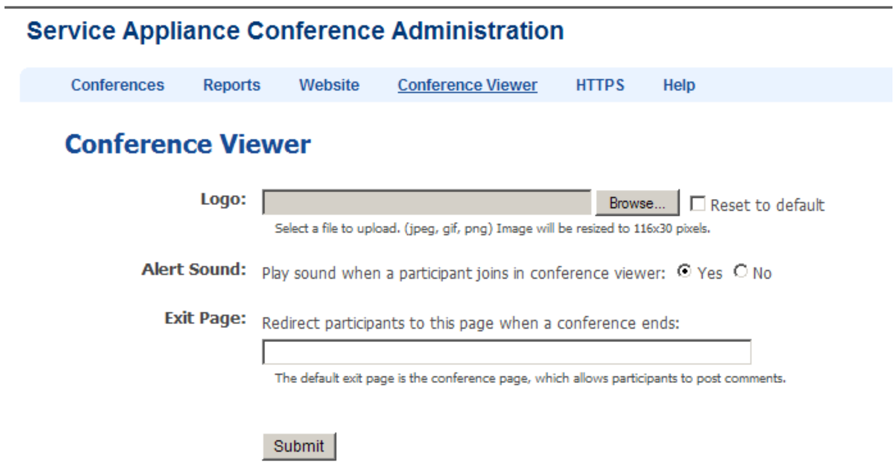


Figure 6-25Conference Viewer Page

Step 5 In the Logo field, enter the path to the logo that you want to appear on the conference viewer page.

Step 6 To have conference sound a tone when a new participant enters, do either of the following in the Alert Sound section:

- Click the **Yes** radio button to activate the tone.
- Click the **No** radio button to deactivate the tone.

Step 7 In the Exit Page field, enter the path to the page that you want to direct participants to when the conference ends.

Step 8 Click **Submit**.

6.8.5 Configuring HTTPS

ShoreTel support HTTPS for secure conference calls. Secure ShoreTel Web conferencing uses 2048-bit encryption. To provide HTTPS security to conference users, you must upload Secure Socket Layer (SSL) certificates or Unified Communication Certificates (UCC) on each SA-100 that you want to provide HTTPS security. This section describes how to obtain an SSL certificate and how to install the certificate on SA-100 units.

6.8.5.1 Obtaining an SSL Certificate

If you already have an SSL certificate, you can install it on your SA-100. See "Uploading an HTTPS Certificate" on page 152 for instructions about installing an SSL certificate. If you do not have an SSL certificate, ShoreTel recommends that you purchase an SSL certificate from a reputable Certificate Authority (CA). To purchase such a certificate, you must first create a Certificate Signing Request (CSR) to present to the CA. You can create a CSR in ShoreWare Director as follows:

Step 1 Launch ShoreWare Director as a user with administrator privilege. (The default administrator, admin, cannot be used.)

Step 2 Click **Administration > Platform Hardware > Voice Switches/Service Appliances > Primary**. The Primary Voice Switches/Service Appliances page appears.

Step 3 Select an SA-100 listing and in the Quick Launch column click **Launch**. The Service Appliance Conference Administration Conferences page appears as shown in Figure 6-26.

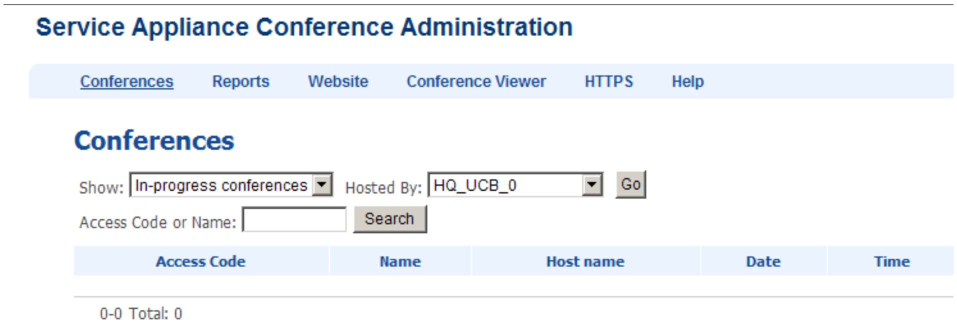


Figure 6-26Service Appliance Conference Administration Conferences Page

Step 4 Click the **HTTPS** tab. The HTTPS tab appears as shown in Figure 6-27.

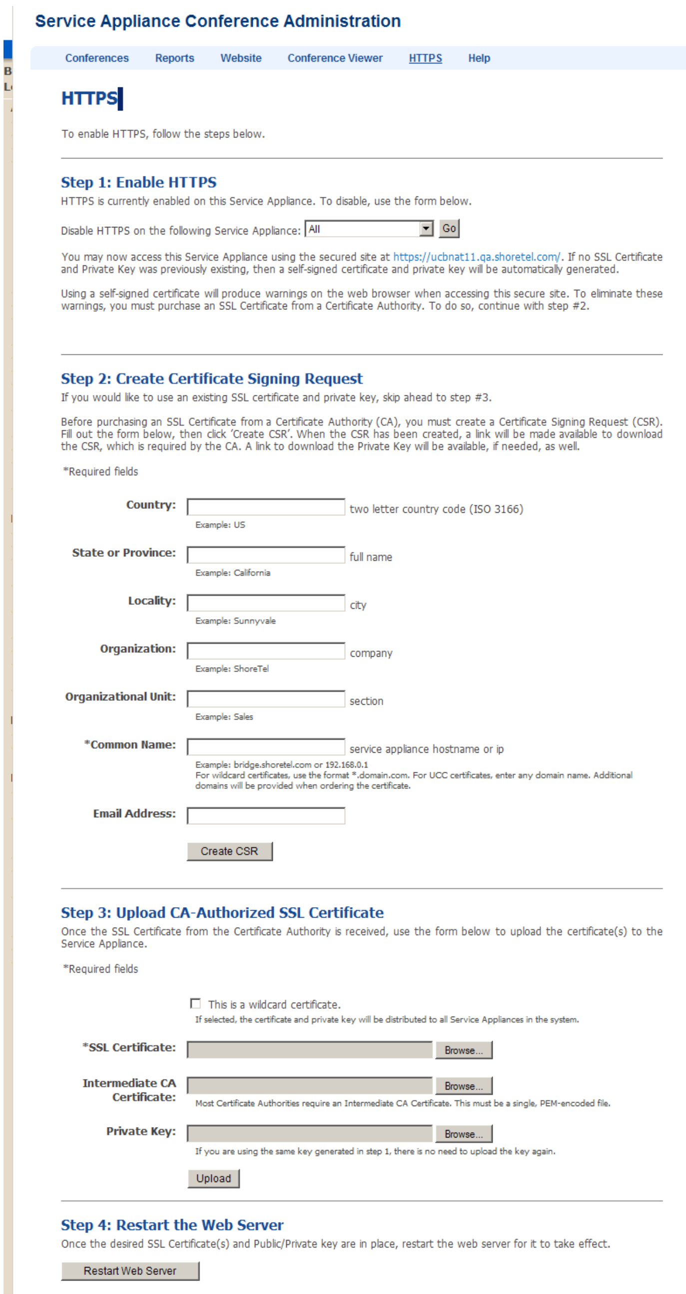


Figure 6-27HTTPS Tab

Step 5 In the respective fields, enter the information listed below. An asterisk (*) indicate that this information is required.

- **Country:** Enter the two-character country code for the country where the system is installed. Refer to ISO 3166 for information about country codes.
- **State or Province:** Enter the name of the state or province where the system is installed.
- **Locality:** Enter the name of the city where the system is installed.

- **Organization:** Enter the legal name of the company or organization or the person that is requesting the certificate.
- **Organizational Unit:** Enter the name of the department that is requesting the certificate.
- ***Common Name:** Enter the Fully Qualified Domain Name (FQDN) for which you want the SSL certificate.

  Note If the CSR is for a wildcard certificate, you must prepend "*." to the common name (e.g. *.domain.com).

  Note If the CSR is for a UCC, enter one only of the domain names for which you want the certification. You will provide additional domain names when you order the certificate.

- **Email Address:** Enter an email address of the administrator who is to handle the certificate.

Step 6 Click **Create CRS**. The ShoreTel system creates a CSR and a private key.

The system generates links that you can use to download the CSR and private key. Download the CSR and private key--if required---and send them to the CA.

6.8.5.2 Uploading an HTTPS Certificate

This section describes how to upload HTTPS certificates. You can upload certificates for a single SA-100 or for all of the SA-100 currently installed on your system depending upon your certificate agreement. To upload HTTPS certificates, do the following:

Note Wildcard certificates that you upload are automatically transferred to all appliances including units that you add to the system later.

Step 1 Launch ShoreWare Director as a user with administrator privilege. (The default administrator, admin, cannot be used.)

Step 2 Click **Administration > Platform Hardware > Voice Switches/Service Appliances > Primary**. The Primary Voice Switches/Service Appliances page appears.

Step 3 Select an SA-100 listing and in the Quick Launch column click **Launch**. The Service Appliance Conference Administration Conferences page appears as shown in Figure 6-26 on page 150.

Step 4 Click the **HTTPS** tab. The HTTPS tab appears as shown in Figure 6-27 on page 151.

Step 5 Scroll to the Upload CA-Authorize SSL Certificate section and do the following:

Step a Check the **This is a wildcard certificate** check box to use the certificate and private key you are about to upload on all SA-100s on your ShoreTel system.

Step b In the SSL Certificate field, enter or browse to the path to the file that contains the SSL certificate that you want to use. This field is required.

Step c In the Intermediate CA Certificate field, enter or browse to the path to the file that contains the intermediate SSL certificate that you want to use.

Step d In the Private Key field, enter or browse to the path to the file that contains the Private Key that you want to use.

Step e Click **Upload**.

Step 6 Click **Restart Web Server**. This reboots the SA-100 and activates all certificates when the unit restarts.

6.8.5.3 Enabling HTTPS

You must have a certificate to run HTTPS. If you already have a certificate, use the process described in "Uploading an HTTPS Certificate" on page 152 to upload it to the appliance. (Wildcard certificates need be uploaded once only. These certificates are automatically transferred to SA-100s already installed on the system and to new SA-100s after they are installed.) If you do not have a certificate, you can obtain one (see "Obtaining an SSL Certificate" on page 149 for instruction about obtaining a certificate) or the system will automatically create a self-signing SSL certificate and private key. A self-signing certificate will allow an SA-100 to create and participate in HTTPS conferences, but as the certificate is not issued by a third party HTTPS security is not optimized. When a self-signing certificate is used, users trying to access the secure conference site will receive a warning message that the security provided is not guarantied.

You can also enable HTTPS on selected SA-100 or on all SA-100 in your system. Because the ShoreTel system uses a distributed approach, we recommends that you enable HTTPS on all of your SA-100s. The distributed approach ShoreTel uses means that multiple SA-100s can be used in a single conference. In conferences where HTTPS is used, SA-100s that are not configured to use HTTPS cannot communicate with SA-100s that are using HTTPS and are not allowed to join.

ShoreTel recommends that if you enable HTTPS on one SA-100 in your environment, that you enable it on all SA-100s in your environment. Mixing HTTPS-enabled SA-100s with SA-100s that do not have HTTPS enabled can cause problems. Users assigned to appliances without HTTPS get an error when they try to access library files stored on an HTTPS-enabled SA-100. The user is able to view the files, but a horizontal blue stripe appears on the right hand side of the pane when they access the files.

This section describes how to enable HTTPS on an SA-100. To enable HTTPS on an SA-100, do the following:

Step 1 Launch ShoreWare Director as a user with administrator privilege. (The default administrator, admin, cannot be used.)

Step 2 Click **Administration > Platform Hardware > Voice Switches/Service Appliances > Primary**. The Primary Voice Switches/Service Appliances page appears.

Step 3 Select an SA-100 listing and in the Quick Launch column click **Launch**. The Service Appliance Conference Administration Conferences page appears as shown in Figure 6-26 on page 150.

Step 4 Click the **HTTPS** tab. The HTTPS tab appears as shown in Figure 6-27 on page 151.

Step 5 In the Enable HTTPS section, make sure that HTTPS is enabled. The function is enabled when the first line in the says the following:

HTTPS is currently enabled on this service appliance.

If the service appliance is disabled, do the following:

Step a In the Enable HTTPS on the following Service Appliance field, select the option that indicates the device on which you want to enable HTTPS.

Note Selecting All enables HTTPS only on the SA-100s that are currently online. The enabling process must be repeated when units that are currently offline are brought online and when new units are added.

Step b Click **Go** to enable HTTPS.

Note When using a self-signed certificate, user uploads to Personal or Shared Library may fail if using the FireFox web browser to access the Conference Web Portal.

Note User's who access the Conference Web Portal on appliances with HTTPS enabled and who use the FireFox web browser must disable Transport Layer Security (TLS) to properly render web pages. To disable TLS in Firefox:

— Select Tools > Options.
— Click the Advanced tab
— Click the Encryption tab.
— Uncheck "Use TLS 1.0".
— Click OK.

CHAPTER 7

Configuring Trunks

This chapter describes how to configure trunks and trunk groups in ShoreTel Director. The topics in this chapter are:

The expanded Trunks link in the navigation frame appears in Figure 7-1.

7.1 Overview

Before beginning, you should understand the different trunk types and trunk features that the ShoreTel system supports.

- A very thorough description of the types of trunks and their associated features is included in the ***ShoreTel 12: Planning and Installation Guide***, Chapter 5, "Trunk Planning and Ordering."
- A detailed description of how the dialing plan, network call routing, and digit manipulation operate is included in the ***ShoreTel 12: Planning and Installation Guide***."

For more information about the features supported outside the U.S. and Canada, refer to the ***ShoreTel 12: Planning and Installation Guide***."

For an overview of the various trunk types and trunk features, refer to the "Trunk Planning and Ordering" chapter in the ***ShoreTel 12: Planning and Installation Guide***.

7.2 Setting Up Trunks

Click **Trunk Groups** under the **Trunks** link in the navigation frame to open the Trunk Groups window, as shown in Figure 7-1.

Trunk Groups — Help

Add new trunk group at site: Headquarters of type: SIP Go

Name	Type	Site	Trunks	DID	Destination	Access Code
Analog Loop Start	Analog Loop Start	Headquarters	0	No	818-5777	9
ATT DID Incoming	Digital Wink Start	Headquarters	8	Yes	818-5777	9
ATT Pri1 Trk Grp	PRI	Headquarters	46	Yes	818-5777	9
ATT T1 2way E&M	Digital Wink Start	Headquarters	8	Yes	818-5777	0
bla T1	Digital Wink Start	Headquarters	0	Yes	818-5777	9
Digital Loop Start	Digital Loop Start	Headquarters	0	No	818-5777	9
Digital Wink Start	Digital Wink Start	Headquarters	8	No	818-5777	9

Figure 7-1 Trunk Groups List Page

The columns in the Trunk Groups list page follow:

- **Name:** This is the name of an existing trunk group. Clicking a name invokes the Trunk Group edit page, where you can edit the trunk group configuration.
- **Type:** This is the type of trunk group.
- **Site:** This is the site of the trunk group. Note that a trunk group cannot span sites. For information about configuring sites, refer to Chapter 3: ShoreTel Sites starting on page 65.
- **Trunks:** This is the number of trunks in the trunk group.
- **DID:** This column shows whether the trunk supports DID. The trunk group supports at least one of the following:
  — DID
  — DNIS
  — Extension
- **Destination:** This is the destination where incoming calls are routed. This may be the auto-attendant, the operator, or an individual user's extension. According to the options you have set, incoming calls are routed, in the order they are listed, to:
  — DNIS
  — DID
  — Extension
  — Destination

  Destination is always available as the last choice for routing.
- **Access Code:** This displays the access code for the trunk group.

7.2.1 Add or Edit a Trunk Group

To add a new trunk group, on the Trunk Groups list page, specify a site from the Add new Trunk Group at site drop-down list, select a trunk group type from the "of type" drop-down list, and click Go. To edit an existing trunk group, click a trunk group from the list of trunk groups in the Name column on the Trunk Groups list page.

When you add a new trunk group or click the name of an existing trunk group in the Trunk Groups list, the applicable Trunk Group edit window appears. Figure 7-2 shows the T1 PRI Trunk Group edit page, which is a superset of all the other Trunk Group edit pages except the E1 Trunk Group edit page and the SIP Trunk Group edit page.

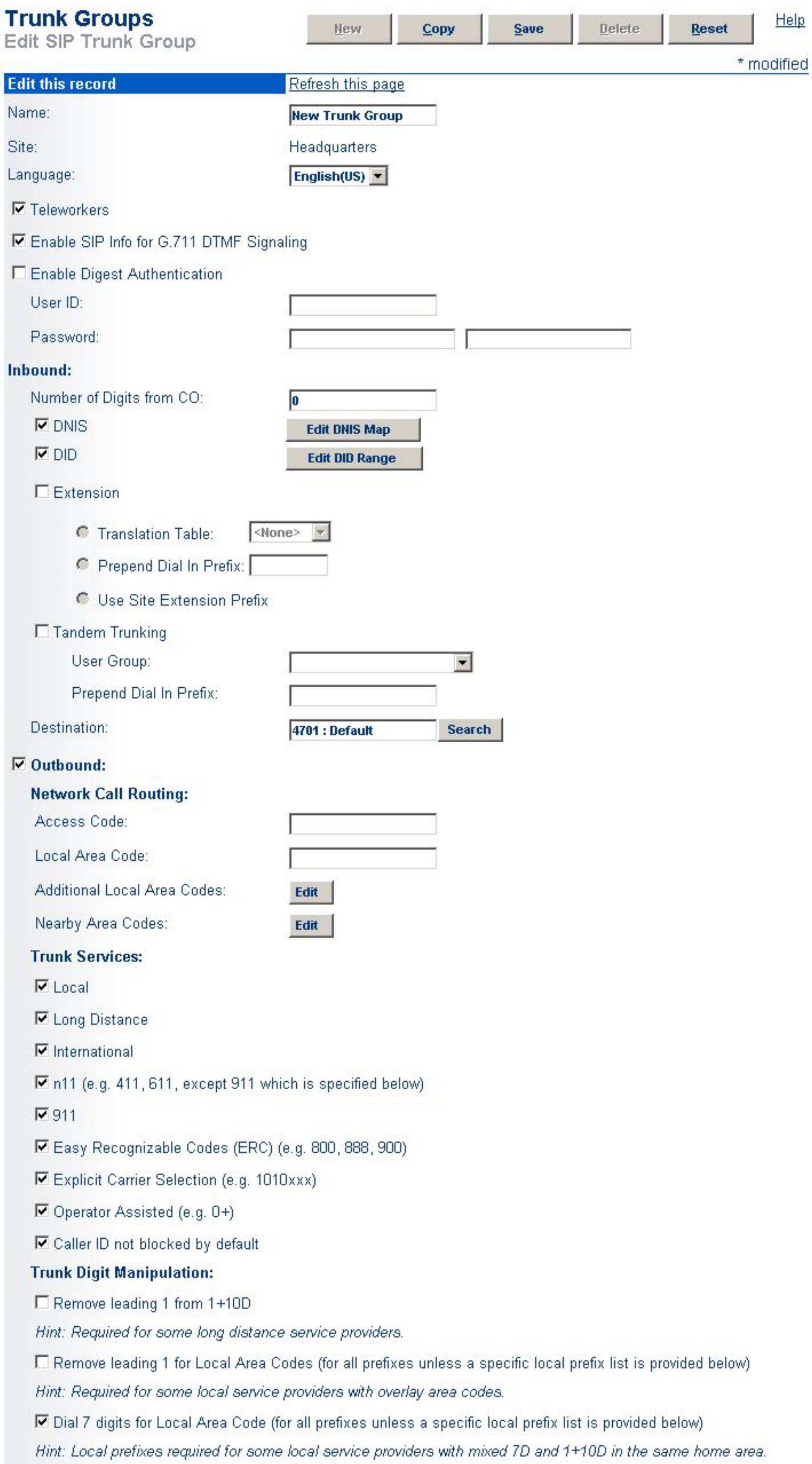


Figure 7-2 Trunk Group Edit Page

The parameters are defined as follows:

- **Name:** This is the name of the new or existing trunk group.
- **Site:** The Name of the site of the trunk group is displayed.
- **Language:** Select the language for the trunk group from the drop-down list.
- **Teleworkers:** This SIP trunk parameter is not used.

  NOTE Third-party SIP devices use the public network for transmitting audio on intersite calls. Intersite bandwidth is used for signaling and call audio.

- **Enable SIP Info for G.711 DTMF Signaling:** This parameter only appears when editing SIP trunk groups.
- **Profile:** Select the desired SIP Trunk Profile. Refer to Section 10.1 for more information.

  Enable this check box when using third-party SIP devices that do not support DTMF negotiation via RFC 2833. If the device does not support RFC 2833 and this check box is not enabled, DTMF negotiation will fail.

- **Enable Digest Authentication:** This parameter only appears when editing SIP trunk groups.

  Enable this check box when using dynamic SIP trunks to provide enhanced security via User ID and Password authentication.

  — **User ID—**This is the user ID associated with the third-party SIP device.

  — **Password—**This is the password associated with the third-party SIP device.

7.2.1.1 Inbound Settings

The **Inbound** settings let you route inbound calls to a single destination, or route calls to a specific destination using DID and DNIS digits.

- **Number of Digits from the CO:** This specifies the maximum number of digits expected from the central office (for User PRI configured trunks).Digit collection terminates when the maximum number of digits are received, when the digit collection time-out is reached, or when an exact match is found.
  Network PRI trunks connected to legacy PBXs collect digits from the legacy PBX side. When the ShoreTel system detects a trunk access code, it ignores the Number of Digits from the CO parameter and routes the call according to the dialing plan.
- **DNIS:** When DNIS is checked, click **Edit DNIS Map** to add or delete entries in the DNIS Map.
- **DID:** When DID is checked, click **Edit DID Range** to add or edit the DID Range as well as view the DID Digit Map.
- **Extension:** If you check this option, calls will route directly to the extension based on the digits received from the central office without any additional configuration. This is very useful when configuring a tie trunk connected to a legacy PBX. Note that the extension length must match the number of digits from the CO.

  — **Translation Table—**When using the On-Net Dialing feature, this parameter allows you to specify a digit translation table that is used to strip one or more digits from calls between two systems with extensions of different lengths.

- **Prepend Dial In Prefix—**When using the On-Net Dialing feature, this parameter allows you to add one or more digits to calls between two systems with extensions of different lengths.
- **Use Site Extension Prefix—**When using the On-Net Dialing feature, this parameter allows you to specify a site extension prefix that will be added to calls from a system that does not have a prefix to another system that does have a prefix.

- **Tandem Trunking:** Tandem trunking allows legacy voice systems to utilize a ShoreTel system for outbound dialing. The ShoreTel system supports network-side PRI, allowing ShoreTel systems to flexibly support digital tie trunks to other systems.
  - **User Group—**Tandem calls are associated with a user group for outbound trunk selection. Inbound calls that are recognized as tandem calls are then redirected to an outbound trunk based on the call permissions and trunk group access associated with the user group set in Director.
  - **Prepend Dial In Prefix—**When needed, you can specify a "dial in prefix" which is pre-pended to digits collected on tandem calls. The concatenated set of digits is then be used in outbound trunk selection for the tandem call.
- **Destination:** The Destination number is mandatory. All inbound calls are routed to a destination, such as an extension (user, workgroup, or route point) or a specific menu. If other destination options have been configured, this destination parameter is the last-choice destination.

  Inbound calls first try to match a DNIS entry, then a DID entry, followed by an Extension entry, and finally Tandem Trunking. If no match is found, the inbound call is routed to the destination you set. If you create a trunk group, the destination is the default auto-attendant.

  An individual trunk group cannot have the overlapping DID or DNIS numbers (received digits).

  Users, Menus, Workgroups, and Route Points can have one DID number but multiple DNIS entries.

7.2.1.2 Outbound Settings

The settings in the Outbound area of the trunk editing window fall into multiple logical groupings. The groups of outbound settings that are defined in this section are Network Call Routing, Trunk Services, Trunk Digit Manipulation, and Local Prefixes. The options for outbound calling are activated by placing a check in the ***Outbound*** check box.

- **Network Call Routing:** The Network Call Routing options are as follows:
  - **Access Code—**Enter the appropriate trunk access code for this trunk group. Typically the access code in the U.S. and Canada is 9.
  - **Local Area Code—**Enter the local area code for this trunk group. This area code is used for Network Call Routing and Digit Manipulation.
  - **Additional Local Area Codes—**Click Edit to enter any additional area codes that are typically associated with overlay area codes. These additional local area codes are used for Network Call Routing and Digit Manipulation.
  - **Nearby Area Codes—**Click Edit to enter area codes that are considered nearby (or "free of cost") for this trunk group. The nearby area codes are used for Network Call Routing.

— **Billing Telephone Number (BTN)—**This function enables the switch to forward an original caller's ID when a received call is redirected by one of ShoreTel's forwarding features. For example, if an outside caller dials a ShoreTel user whose Find-Me setup specifies a cell phone, the cell phone can show who dialed the ShoreTel number. The applicable forwarding features are Find Me, some of the Call Handling Modes, PSTN Failover, Office Anywhere, Allow Additional Phones to Ring Simultaneously, and Extension Reassignment. This role for BTN is described under the next heading, Trunk Services, and in the "Purpose of the Billing Telephone Number for Caller ID" on page 175.

- **Trunk Services:** This area of the Outbound settings in the Trunk Groups editing window lets the system administrator configure the trunk group for the outbound trunk services described in subsequent bullet items. If an option is not enabled, the trunk group does not provide that service (unless the text states otherwise).

— **Local—**Select this check box to enable local calls.

— **Long Distance—**Select this check box to enable long-distance calls.

— **National Mobile** (not shown)**—** This check box only appears for PRI trunks in countries with "caller pays" billing plans (e.g. Ireland). Select this check box to allow users to call mobile numbers. Clear the checkbox if you do not want to incur the associated costs of allowing users to call mobile phones in "caller pays" environments.

— **International—**Select this check box to enable international calls.

— **Enable Original Caller Information—**This function enables the switch to forward an original caller's ID when a received call is redirected by one of ShoreTel's forwarding features. For example, if an outside caller dials a ShoreTel user whose Find-Me setup specifies a cell phone, the cell phone can show who dialed the ShoreTel number. The applicable forwarding features are Find Me, some of the Call Handling Modes, PSTN Failover, Office Anywhere, Allow Additional Phones to Ring Simultaneously, and Extension Reassignment.

This enable is a starting point for other steps that are necessary to transmit the original caller ID. For descriptions of the tasks that follow after Enable Original Caller Information is enabled, see the following:

  - For details about forwarding the original caller ID, see the "Forwarding Original Caller ID Outside a ShoreTel Network" on page 174.
  - For the class of service (COS) that a user must have to ensure that the forwarding of an outside call is permitted, see the "Classes of Service" on page 314.
  - The function Send Incoming Caller ID must be enabled for call forwarding features such as Find-Me, External Assignment, and Allow Additional Phones. This enable is illustrated in Chapter 12, Configuring User Features. Refer to the description in Find Me and External Assignment subsection of the "Forwarding Original Caller ID Outside a ShoreTel Network" on page 174.
  - To respond to carriers who do not validate the caller ID in a SETUP message for an original caller ID, see the "ISDN Profile for RNIE" on page 187. This advanced (and rarely needed) task follows the tasks described in the "Forwarding Original Caller ID Outside a ShoreTel Network" on page 174.

IMPORTANT: In ShoreTel Releases 11, 10.2, 10.1, and 10, the forwarding of the original caller ID to an outside device relied on custom dial plan elements from

ShoreTel TAC. If TAC implemented such a custom dial plan, TAC must remove the elements related to this function before the current capability can work. (This issue does not exist for customers whose *new* installation contains the present implementation described in this book.) If a system with such a dial plan is upgraded, problem behaviors related to call forwarding or original caller ID can show up. Some possible behaviors are:

- Forwarded calls go to the user's voicemail instead of out the trunk.
- Forwarded calls are rejected by the carrier.

Upgraded customers who know or suspect that such a plan has been used should contact TAC for help. However, some customers might not know that a custom dial plan has been used for original caller ID and should, therefore, monitor the call forwarding and caller ID performance after an upgrade.

— **n11** (for example, 411 or 611, but not 911—Select this check box to enable telephone service calls (for services such as directory assistance or repair service).

— **911—**Select this check box to enable emergency 911 calls. To support 911 in the U.S., at least one trunk group per site must allow 911 calls. For a detailed description of 911 support, see Appendix , starting on page 595.

— **Easy Recognizable Codes (ERC)—**Click this check box to enable services such as toll-free dialing for easily recognized codes like 800, 888, or 900.

— **Explicit Carrier Selection—**Click this check box to specify a particular long-distance carrier. The format of the access code is 1010*xxx*. An example is 1010811.

— **Operator Assisted** (for example, 0+)**—**Click this check box to enable the trunk group to dial the operator.

— **Caller ID not blocked by default—**Click this check box to pass Caller ID information by default on outbound calls. To block all calls, clear this option. In the United States, the user can override this option with Vertical Service Codes.

- **Trunk Digit Manipulation:** This parameter lets you control how the trunk group manipulates the telephone number before outpulsing the digits to the central office.

  All North American numbers are converted into 1+10-digit format internally before being passed to the trunk group for digit manipulation.

  — **Remove leading 1 from 1+10D—**Click this check box to drop the leading "1". Dialing only ten digits is required by some long-distance service providers.

  — **Remove leading 1 for Local Area Codes** (for all prefixes unless a specific local prefix list is provided below)**—**Click this check box to drop the leading "1" for the local area codes (Local and Additional Local). Dialing only ten digits for local area codes, particularly with overlay area codes, is required by some local service providers. If a local prefix list provided, the leading "1" is removed for the all entries in the list.

  — **Dial 7 digits for Local Area Code** (for all prefixes unless a specific local prefix list is provided below)—Click this box to enable the trunk to dial numbers in the local area code with seven digits. Some local service providers require this enable.

  If a local prefix list provided, seven digits are dialed for all entries in the list (Local Area Code only, not Additional Local Area Codes).

- **Local Prefixes:** Click the **Go to Local Prefixes List** link to view, add, or edit the local prefixes for the site. When a local prefix list is used, prefixes that are not in the prefix list are considered "long distance" and require a long distance trunk service. Local prefixes can be importaed from a CSV file. For more information, see "Importing Local Prefixes" on page 162.

- **Prepend this Dial Out Prefix:** Select a prefix from the drop-down list. It is pre-pended to the dial-out string that results from the other rules. (The Dial Out Prefix is not applied to Off-System Extension calls.) A dial-out prefix is typically required when connecting to, and leveraging the trunks on, a legacy PBX.
- **Off-System Extensions:** Click **Edit** to add or edit extensions ranges that can be accessed through this trunk group. This is typically used for setting up a tie trunk to a legacy PBX and configuring coordinated extension dialing. See also "Escalation Profiles and Other Mailbox Options" on page 448.

  The Dial Out Prefix and Digit Manipulation rules are not applied to Off-System Extensions.
- **Translation Table:** Specify the digit translation table that is used to strip one or more digits from calls between two systems with extensions of different lengths.

  ShoreTel does not apply inbound digit treatment to digit strings beginning with a trunk access code (such as 9) as would occur in tie trunk configurations. Digit strings beginning with a trunk access code are routed according to the dial plan.

  A user group named Account Codes Service is created for use by the Account Codes Service. It is not available for assignment to users but it may be edited.

7.2.2 Importing Local Prefixes

You can import or export local prefixes in CSV format. Local prefix lists can be purchased or obtained free from various web sites.

To import local prefixes from a CSV file:

Step 1 Navigate to the Trunk Group profile for which you want to import local prefixes.

Step 2 Scroll to the Local Prefix section at the bottom of the page and click **Go to Local Prefixes List**. The Local Prefix page appears.

Step 3 Click **Add New List**. The Edit Local Prefix page appears as shown in Figure 7-3.

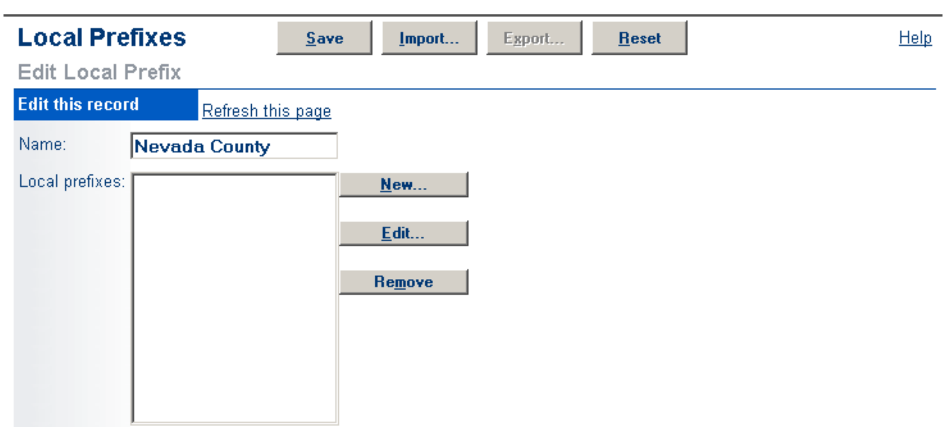


Figure 7-3 Local Prefixes Edit Page

Step 4 Click the **Import** button. The Import Local Prefixes dialog box appears as shown in Figure 7-4.

Figure 7-4 Local Prefixes Import Dialog Box

Step 5 In the field, enter the path and name of the CSV file that you want to import or click **Browse** to search for the file.

Step 6 Click **Upload**. The Local Prefixes edit page appears as shown in Figure 7-5.

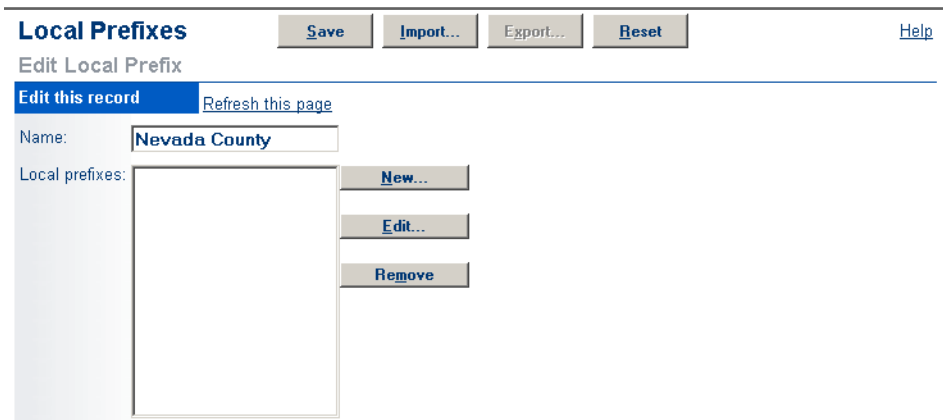


Figure 7-5 Local Prefixes Edit Page, Populate

Step 7 Edit the list as needed. You can re-name the list and add, edit, or remove prefixes.

Step 8 Click **Save**. The list you created is now available from the Local Prefix drop-down list.

7.2.3 Exporting a Local Prefix List

To export a local prefix list:

Step 1 Navigate to the Trunk Group profile that contains the list that you want to export.

Step 2 Scroll to the Local Prefix section at the bottom of the page and click **Go to Local Prefixes List**. The Local Prefix page appears as shown Figure 7-6.

Figure 7-6 Populated Local Prefix List

Step 3 Click the list you want to export. The Local Prefixes edit page appears as shown in Figure 7-7.

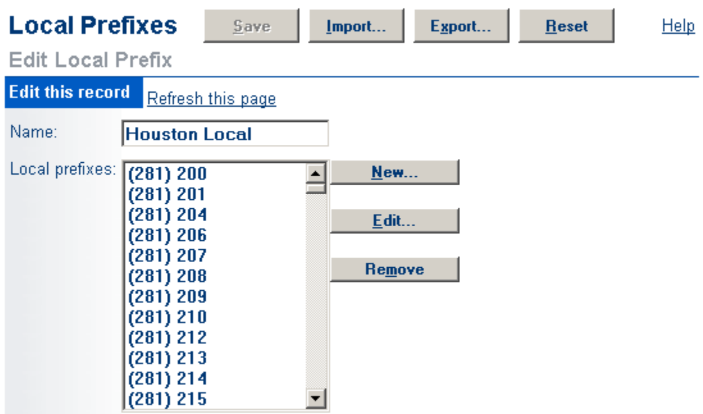


Figure 7-7 Populate Edit Local Prefix Dialog Box

Step 4 Click the **Export** button. Your web browser opens the file in a new window.

Step 5 Specify where you want the file and save it as a text file.

7.2.4 DID Ranges

Direct Inward Dialing (DID) is a feature offered by telephone companies for use with their customers' PBX systems, where the telephone company allocates a range of numbers to a customer's PBX. As calls are presented to the PBX, the number that the caller dialed is also given, allowing the PBX to route the call to the intended party.

A DID range is a list of consecutive (non-overlapping) DID numbers assigned to a Trunk Group. A DID number can be assigned to multiple trunk groups.

Available DID **numbers** are DID numbers within a range that are not assigned to a user or entity within the context of that range. DID number availability within a range does not consider DNIS assignments. Numbers assigned as a DNIS number are still enumerated as available within a DID range; attempts to assign these DID numbers will be unsuccessful.

DID assignment field and range indication fields are listed on the User, Workgroup, Route Point, Menu, Hunt Group, and Bridged Call Appearance pages.

7.2.4.1 Creating a DID Range

To set a DID range, click Edit DID Range from the appropriate Edit Trunk Group page to invoke the DID Range page as shown in Figure 7-8.

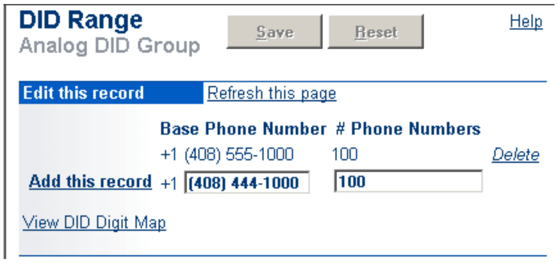


Figure 7-8 DID Range Page

For each block of DID numbers, enter the base phone number and the number of phone numbers supplied by that block in the appropriate fields, and click “Add this record.” When finished, click Save. You can configure multiple ranges per trunk group.

To view the currently configured DID Digit Map (Figure 7-9), click View DID Digit Map. From here, you can view pertinent information about each DID number in the ShoreTel system.

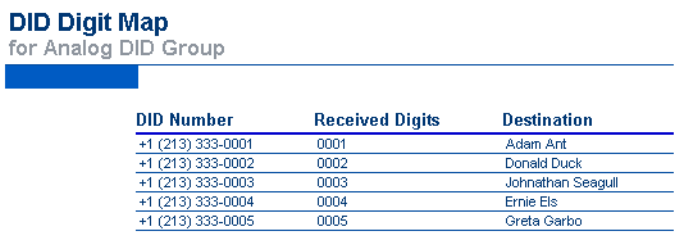


Figure 7-9 DID Digit Map

7.2.4.2 Assigning DID Numbers from a Range

DID assignment sections on Director pages consist of two parameters: DID Range and DID Number. Figure 7-10 displays these parameters on the Edit User page. Other affected pages are constructed similarly.

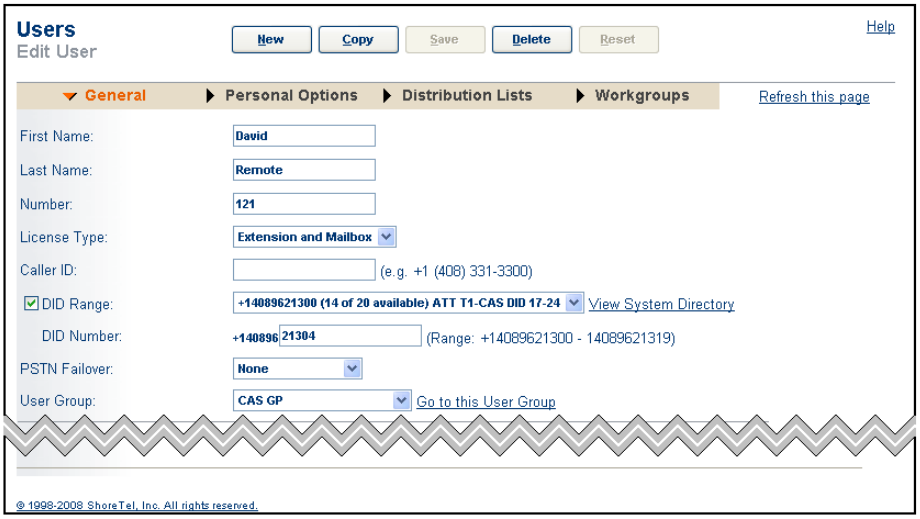


Figure 7-10DID Parameters on Edit User Page

- The DID Range parameter displays the complete base phone number and trunk group name.
  - — To authorize the user to use a DID number, select the checkbox located left of the DID Range text.
  - — Access the drop-down menu to specify the DID Range from which the entity's DID number will be selected. Each DID Range corresponds to a trunk group and lists the number of available numbers, as shown in Figure 7-11.

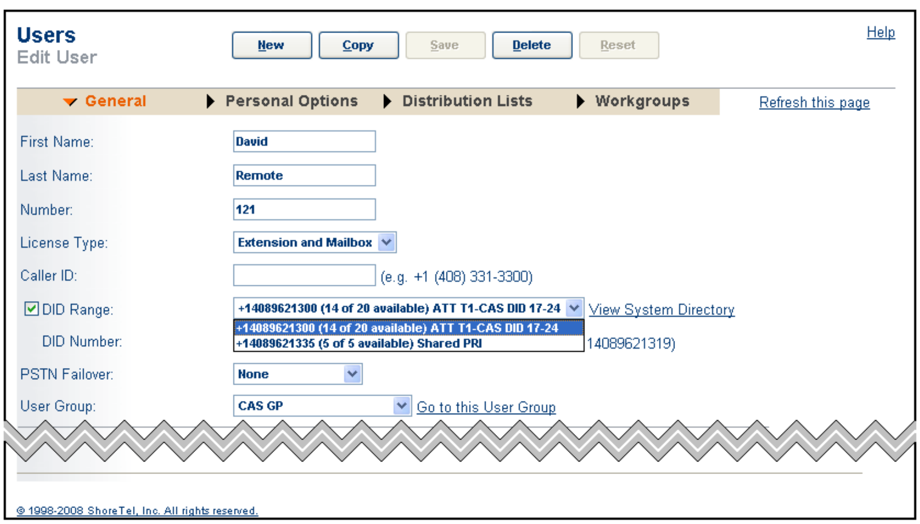


Figure 7-11DID Range parameter – drop-down menu

- A link to the System Directory page is located right of the DID Range data entry field.

  Administrators can assign a DID number by viewing the System Directory page to find an available DID number, then selecting the number from the drop-down list. Unassigned numbers within a DID range may not be available if the number is assigned as a DNIS in the same trunk group.

- The DID Number parameter assigns the specified DID number to the entity. The prefix located left of the data entry field and the range located right of the data entry field are based on the selected DID Range.

7.2.5 Edit DNIS Digit Map

To edit the DNIS Digit Map, do the following:

Step 1 Navigate to the Trunk Group that you want to support DNIS.

Step 2 Click the **Edit DNIS Map** button. The DNIS Digit Map page appears (Figure 7-12).

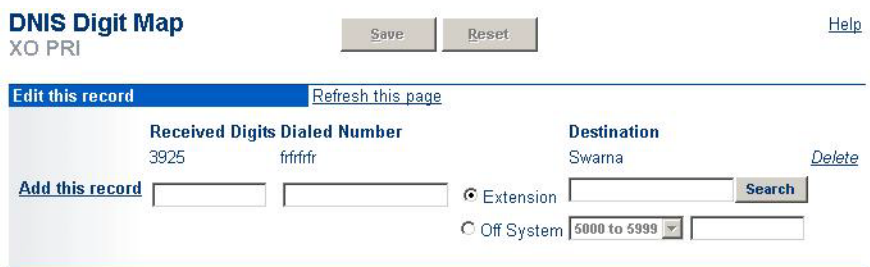


Figure 7-12DNIS Digit Map Edit Page

Step 3 In the **Add this record** field in the Received Digits column, type the DNIS number that the Telco sends.

Step 4 Do one of the following to specify the extension to which the DNIS is routed:

- To map the DNIS to an internal extension, click the **Extension** radio button and enter the extension you want to use in the Destination field. (You can click the Search button to view the list of available extensions.)
- To map the DNIS to an off-system extension, click the **Off-System** radio button, select the extension range that includes the extension you want to use, and enter the extension you want to use in the Destination field.

Step 5 To create an identifier for the map, type a description in the field in the Dialed Number column. The description can include up to 26 alpha and numeric characters. This description appears to call recipients and in call detailed reports (CDRs).

Step 6 Select a Destination extension by using the Search button and then click **Add this record**.

7.2.6 Additional Local Area Codes

To configure additional local area codes, click the associated button on the Trunk Group edit page. The Addition Local Area Codes dialog box appears (Figure 7-13).

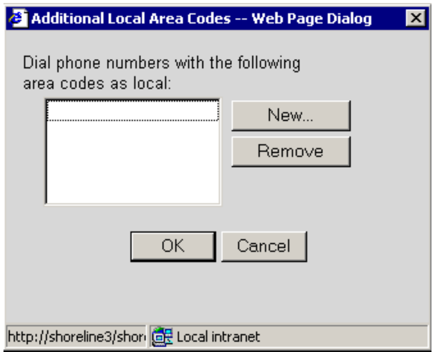


Figure 7-13Additional Local Area Codes Dialog

Click **New** to add a new additional local area code and Remove to delete one.

7.2.7 Nearby Area Codes

To configure nearby area codes, click the associated button on the Trunk Group edit page (see Figure 7-2 on page 157). The Nearby Area Codes dialog box appears as shown in Figure 7-14.

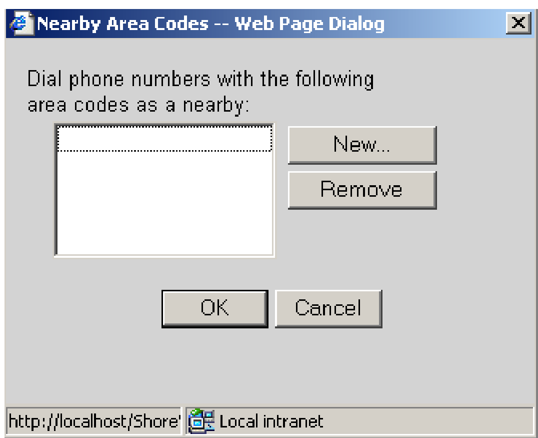


Figure 7-14Nearby Area Codes Dialog

Click **New** to add a nearby area code or **Remove** to delete one.

7.2.8 Prefix Exceptions

To configure the local prefixes list, click the link Go to Local Prefixes List located near the bottom of the Trunk Group edit page. The Local Prefixes dialog box is presented.

If you click Delete, changes take place immediately.

To configure local prefixes, click “add new list” or an existing list to bring up the Local Prefixes dialog box as shown in Figure 7-15.

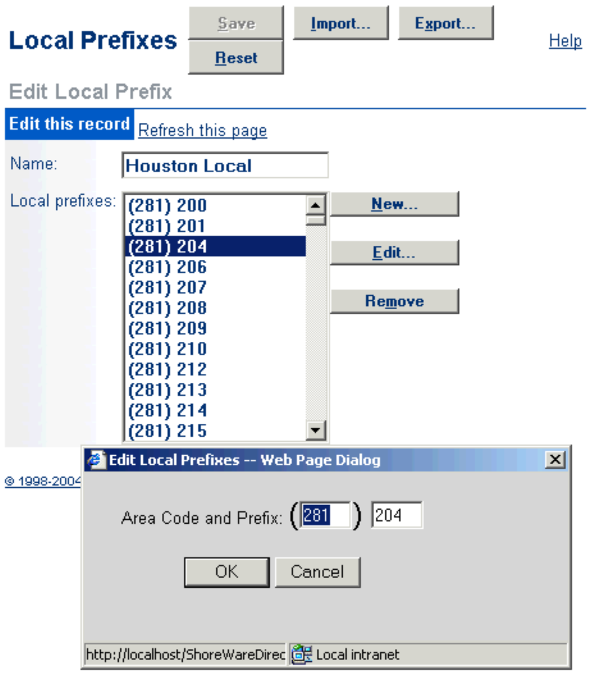


Figure 7-15Local Prefixes Dialog Box

Changes take place when you click *OK* within the Local Prefixes dialog.

7.2.9 Off-System Extensions

If you are using off-system extensions, you can list them by clicking the Edit button on the Trunk Group edit page. Off-system extensions are typically used when setting up a tie trunk to a legacy PBX and configuring coordinated extension dialing. Also refer to Section 13.3.1. Figure 7-16 shows the dialog boxes used to define ranges of extensions.

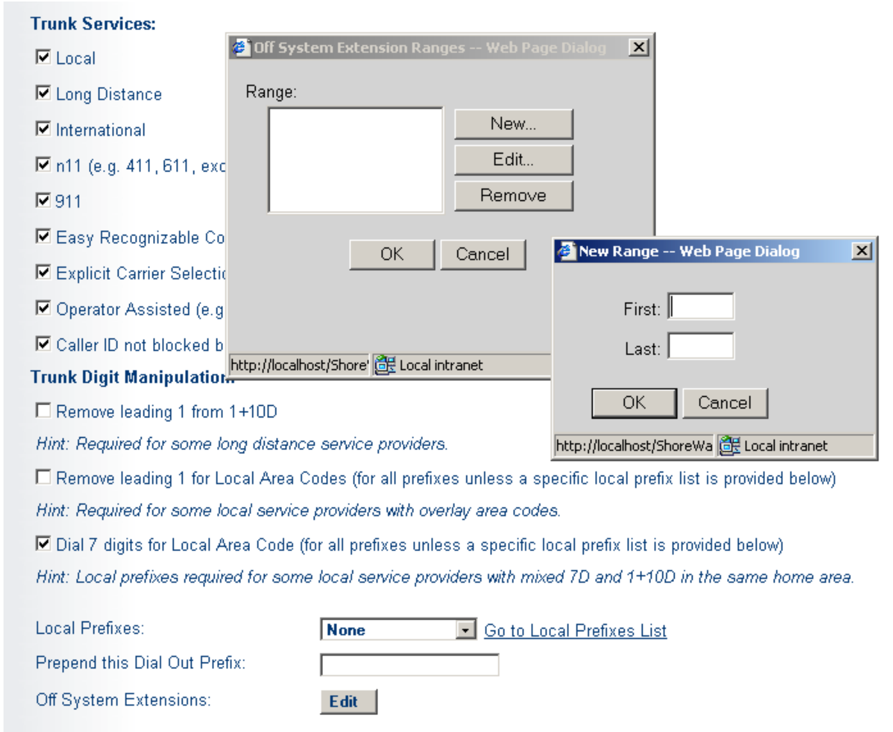


Figure 7-16Configuring Off-System Extensions

7.2.10 Tandem Trunking

Tandem trunking treats digits on an incoming trunk call as a PSTN number. Received digits are tested against DNIS, DID, Extension, and Tandem Trunking, in that order. When Tandem Trunking is enabled, the number of digits from the CO may have no effect if the first digit(s) matches a Trunk Access Code. To define trunk access and call permissions, associate a user group with the tandem trunk group.

Any Dial In Prefix is pre-pended to each set of inbound digits. You can use DNIS/DID/ Extension matching with a Dial In Prefix.

When using NI-2 signaling on PRI trunks—for example in a tie trunk scenario—Caller ID name is also captured, when available, on all inbound calls. For outbound calls, the Caller ID name is delivered for calls that are made to off-system extensions, but not generally for all outbound calls.

Tandem calls are reported in the Trunk Detail and Trunk Summary reports, with incoming and outgoing legs reported according to the reports' formats.

7.2.11 Centrex Flash

Centrex Flash can be programmed on a custom button so that a user can transfer a call to another number. When the user presses the custom button, a flash is generated on the current call, the central office presents dial tone, and the user can dial any PSTN number. When ring-back is heard, the user can hang up the handset to complete the transfer.

Note that the user is connected directly to the central office, so no access code is required, no permissions are checked, no account code is supported, and no CDR logging of the second call occurs.

Centrex Flash is useful in branch offices or small office environments with a limited number of analog Centrex lines. If an external caller needs to be transferred to an external number, the two trunks will be cleared, instead of quickly busying out the trunks after a few transfers. In this manner, the feature reduces the number of physical trunks needed to transfer calls, since no trunks are in use after the transfer is completed.

Centrex transfer is supported only on analog loop-start trunks. If the call is not on an analog loop-start trunk, the operation will have no effect. The trunk on which the call exists must be configured on one of the following switches:

- ShoreTel Voice Switch 40/8
- ShoreTel Voice Switch 60/12
- ShoreTel Voice Switch 120/24
- ShoreTel Voice Switch-E1
- ShoreTel Voice Switch-T1

The feature replaces a trunk-to-trunk transfer in which two trunks are tied up for the duration of the call. The current call must be connected and also must be a two-party call.

For information on configuring Centrex Flash, please see the "Programmable IP Phone Buttons" on page 211.

7.2.12 Pause in Dialing – Trunk Access

The Pause in Dialing – Trunk Access feature allows the insertion of commas (,) into Prepend Dial Out prefix of a trunk group to specify one-second pause periods during the transmission of pulse digits. Pause periods are permitted in the following trunk group types that send digits as pulses:

- Analog Loop Start
- Digital Loop Start
- Digital Wink Start

Example: Assume the prepend dial-out prefix of an analog trunk group is “9,,”. When a user dials “914085551111”, the following pulse-digit sequence is transmitted on the trunk: “9<*silence for two seconds*>14085551111”.

Commas are permitted anywhere within the Prepend Dial Out Prefix.

Feature restrictions include:

- The pause cannot be used to insert account codes.
- Commas in Communicator dial strings are ignored,.
- Commas in dial strings sent by other TAPI applications are ignored.

To enter a dial pause for a specific trunk group, perform the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration -> Trunks -> Trunk Groups**.

Step 3 Select the trunk group for which you want to introduce a dial pause. The Trunk Groups Edit page for the selected group opens, as shown in Figure 7-17.

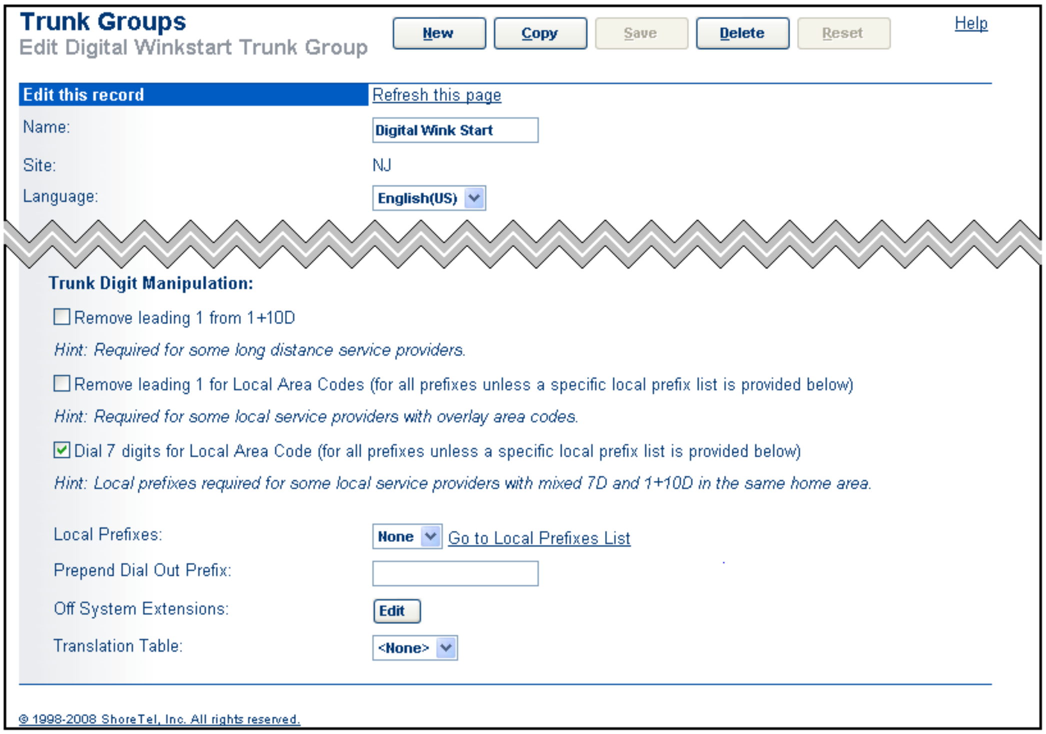


Figure 7-17Modifying the Prepend Dial Out Prefix Setting

Step 4 Enter the code in the Prepend Dial Out Prefix data entry field located near the bottom of the page.

Step 5 Press the Save button at the top of the page.

7.3 Individual Trunks

This section explains how to configure individual trunks once you have created the associated trunk groups as described in "Setting Up Trunks" on page 155. To create a trunk profile, do the following:

To invoke the Trunks by Group page, click Trunks from the navigation frame, and then click Individual Trunks. The Trunks by Group page appears, as shown in Figure 7-18.

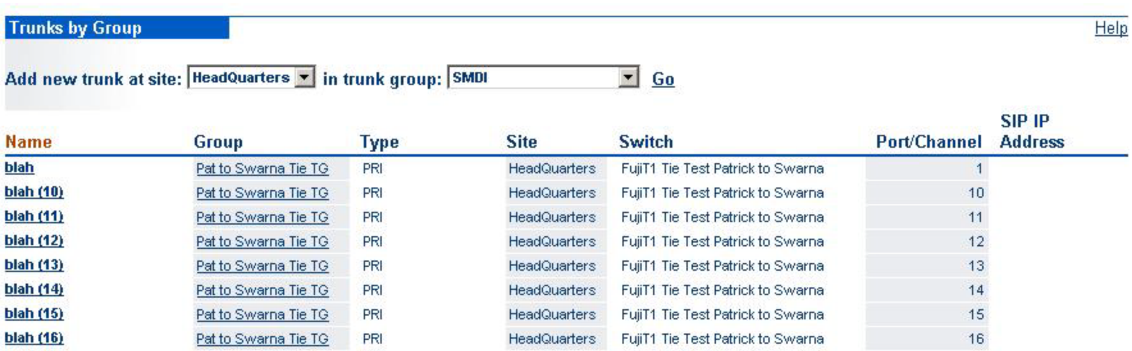


Figure 7-18Trunks by Group Page

To select a trunk group, select a site from the “Add new trunk at site” drop-down list, then select a trunk group from the “In trunk group” drop-down list, and click Go. The trunks belonging to the selected trunk group appear. The columns in the Trunks by Group page are as follows:

- **Name:** This is the name of an individual trunk in the group. Click an entry in the ***Name*** column to invoke the Trunks edit page to edit the individual trunk’s parameters.
- **Group**: This is the trunk group name. Click a group name to invoke the Trunk Group edit page and edit the trunk group’s parameters.
- **Type**: This ID can range from 1 - 6 and corresponds to the trunk type (e.g. analog DID, analog loop start, SIP, etc.)
- **Site:** This is the location of the trunk and can be the Headquarters location or one of the remote locations.
- **Switch:** This is the IP host name of the ShoreTel voice switch to which the individual trunk is connected.
- **Port/Channel:** This is the port number or channel to which the individual trunk is connected.
- **SIP IP Address**: This IP address only applies to SIP trunks and corresponds to the SIP end point device.

7.3.1 Add or Edit a Trunk

Whether you choose to add a new trunk or edit an existing trunk, the Trunks edit page appears.

7.3.1.1 Non SIP Trunk Parameters

The parameters on the Trunks edit page for non-SIP trunk shown in Figure 7-19 are described below.

Figure 7-19Trunks Edit Page

- **Site:** This is the name of the site at which the trunk and trunk group are located.
- **Trunk Group**: This is the name of the trunk group to which the trunk group belongs. You cannot change the name of the trunk group on this page.
- **Name**: This text-entry field lets you enter the name of the individual trunk.

- **Switch Channel:** This drop-down list selects the channel to which this trunk connects.
- **Jack #:** This is the patch-panel jack number that is associated with the trunk's switch port. This is an optional parameter.

7.3.1.2 SIP Trunk Parameters

This section describes the parameters in the Trunks edit page for SIP trunks (Figure 7-20).

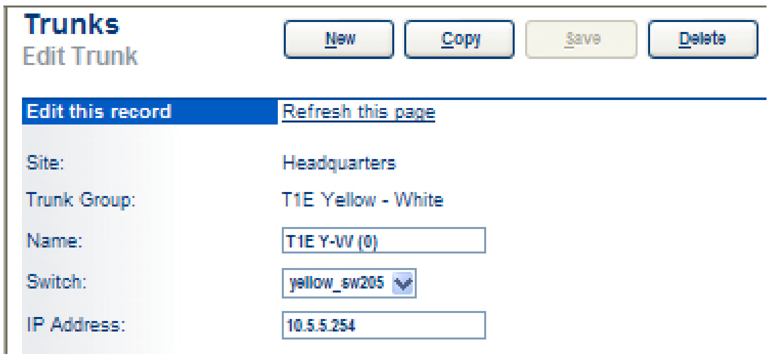


Figure 7-20SIP Trunks Edit Page

- **Site:** This is the name of the site at which the trunk and trunk group are located.
- **Trunk Group:** This is the name of the trunk group to which the trunk group belongs. This field is informational and cannot be changed on this page.
- **Name:** This text-entry field lets you enter the name of the individual trunk.
- **Switch:** This drop-down list lets you select the switch to which this trunk connects.
- **IP Address:** The IP address of the switch.

7.4 Forwarding Original Caller ID Outside a ShoreTel Network

This section describes the configuration tasks for ensuring that the original caller ID goes out an ISDN trunk when a call to a ShoreTel phone is forwarded outside the ShoreTel deployment. This section applies to a trunk group regardless of the way that carriers and service providers (carriers for short) validate caller ID, but it also describes some of the operational behaviors related to forwarding the original caller ID and the different ways that carriers validate caller ID.

When a ShoreTel switch forwards a call out a trunk, the Q.931 SETUP message contains information about the original caller. However, the carriers are not uniform in the way they validate caller IDs and might not be clear to the ShoreTel customer. ShoreTel's response to these differences are described in this section.

In general, this section focuses on:

- Enabling the Enable Original Caller Information function.

- After Enable Original Caller Information is enabled on a trunk group, other capabilities are enabled or requirements activated (such as “Send Incoming Caller ID” in an individual user’s configuration).
- Reasons for using the Billing Telephone Number field for a trunk group.

7.4.1 When Carriers Validate the Caller ID

Many carriers validate the caller ID to ensure that the number is within the range of DID numbers that they have on record for the ShoreTel customer. If the number is outside the range, a carrier could reject the call but usually checks the redirecting number field to find a number within the DID range.

If a call enters a ShoreTel network and is then forwarded out an ISDN trunk to a remote device, the number of the forwarded caller can be outside the DID range on record. If the caller ID is outside the DID range, the carrier can check the content of the Redirecting Number Information Element (RNIE) to see if the number that forwarded the call is within the DID range. The redirecting number belongs to ShoreTel user whose phone forwarded (redirected) the original call. Therefore, the contents of the RNIE field can match the carrier’s records. The result is that the call is forwarded, and the far end device can display the ID of the original caller.

Although most carriers that verify the RNIE inevitably send the intended caller ID, some carriers automatically forward the RNIE contents instead of the original caller ID. In this situation, the original caller ID does not reach the outside terminating device.

7.4.2 When a Carrier Does Not Validate the Caller ID

Some carriers and service providers do not validate the caller ID, so they do not determine what to provide at the destination phone. Therefore, a ShoreTel user who is remote and wants to see the original caller ID might not see the caller ID even with the correct configuration on the ShoreTel Voice Switch. To address this uncertainty, ShoreTel supports an *RNIE ISDN profile* that determines what the carrier should display (even though the provider does not validate the caller ID). For a description of this mechanism, see the ISDN Profile for Redirecting Number Information Element section.

7.4.3 Purpose of the Billing Telephone Number for Caller ID

In general, the billing telephone number (BTN) is used by carriers for billing a ShoreTel customer. In contrast, a different role exists for the Billing Telephone Number field in the Trunk Group configuration. This field is not for billing purposes but rather for supporting an alternative to individual user DIDs when an alternate is needed. For example, for a switch to forward the original caller ID, the ShoreTel user who redirects the call outside must have a DID and a Caller ID entry (in the Individual Users edit window) that fit within the trunk group’s DID range. However, for a variety of reasons, a user might not have a DID or a number within the trunk’s DID range. For example, a ShoreTel user at a remote site would have a telephone number that is outside the range at the location of the server. In this case, the contents of the Billing Telephone Number field go in the RNIE space instead of that user’s DID number.

The Enable Original Caller Information checkbox and the Billing Telephone Number field are in the Trunk Groups page shown in Figure 7-21. The Billing Telephone Number field becomes active and is auto-filled with the base number of the trunk group’s DID range by default when the Enable Original Caller Information checkbox is marked.

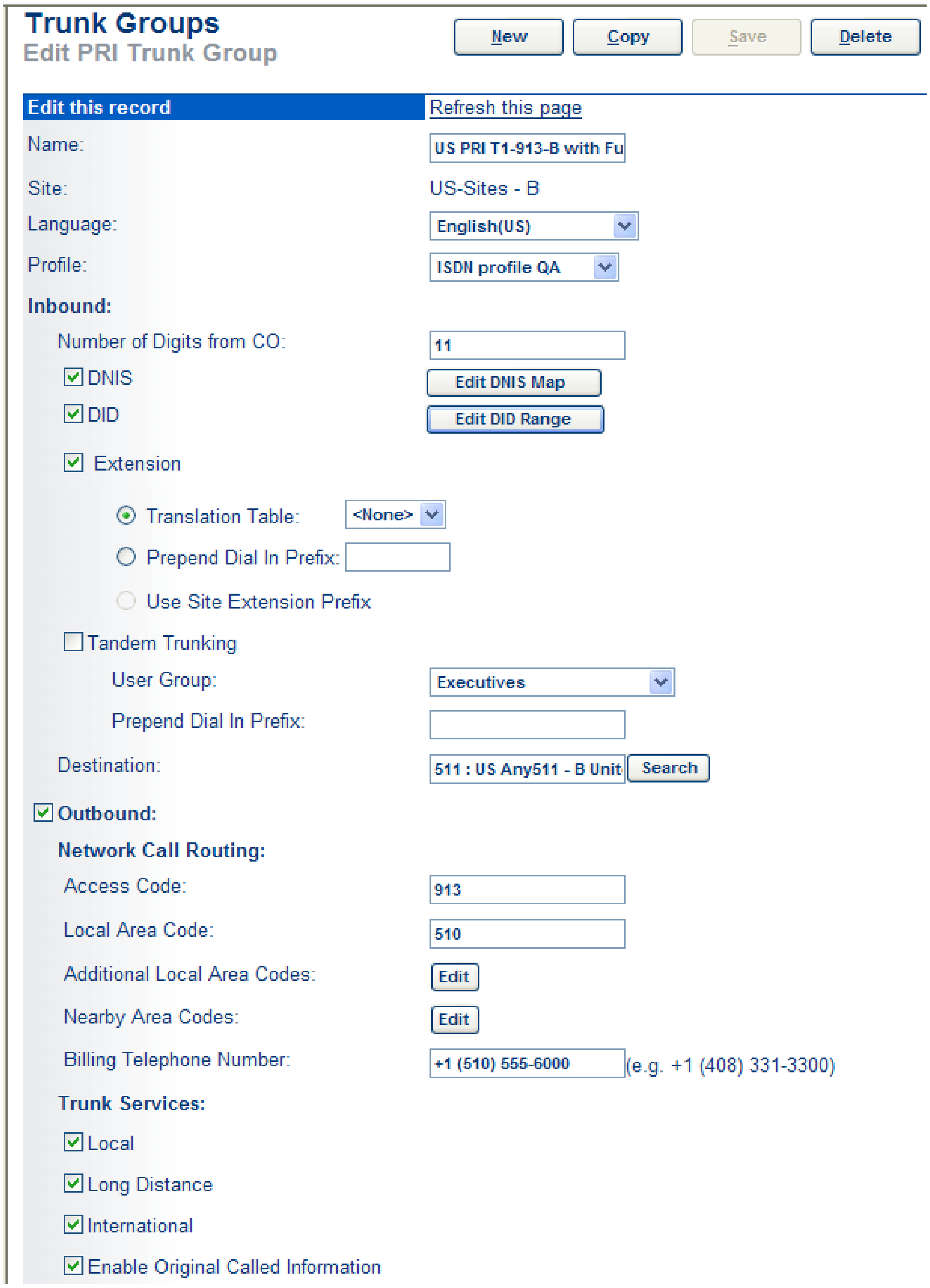


Figure 7-21Trunk Groups Editing Window

A Q.931 SETUP message contains information elements for the caller ID and a redirecting number. The carrier normally finds the caller ID to be within the DID range for the trunk. If the caller ID is outside the range, the provider checks the RNIE field to determine whether the number that redirected the call is within the DID range. If the ShoreTel user does not have a DID to serve as the caller ID, the Billing Telephone Number field in the Trunk Groups window can provide the redirecting number. This field can contain one of several types of phone numbers.

The preferable order of values to have in the Billing Telephone Number field is as follows:

1. The first number in the trunk group's DID range (the default).
2. The actual billing telephone number of the ShoreTel customer (and used by the carrier for billing purposes), typed into the field by the system administrator.
3. The CESID (if the trunk group is configure to support CESID).

The Billing Telephone Number field is activated and populated with the base number of the DID range when the Enable Original Caller Information function is enabled.

NOTE In Release 11.1, the Billing Telephone Number field was added to the Trunk Groups editing window. Some customers who upgrade might not know about this field. Regardless, the field is automatically populated with the base number of the DID range when the original caller ID function is enabled.

7.4.4 Important Issue with Early Implementations of Original Caller ID

This section can be skipped for *new* installations of Release 11.2, 12.1, 12.2, and so on, because the issue does not exist for new installations.

Before the current release, implementation of Original Called Information relied on a custom dial plan. When a ShoreTel deployment with this custom dial plan is upgraded to Release 11.2, 12.1, 12.2, and so on, the ShoreTel TAC must remove such a custom dial plan because it interferes with the current function.

Upgraded customers who know or suspect that such a plan has been used should contact TAC for help. However, some customers might not know that a custom dial plan has been used for original caller ID and should, therefore, monitor the call forwarding and caller ID performance after an upgrade.

If a system with such a dial plan is upgraded, problem behaviors related to call forwarding or original caller ID can show up. Some possible behaviors are:

- Forwarded calls go to the ShoreTel user's voicemail instead of out the trunk.
- Forwarded calls are rejected by the carrier.

7.4.5 Enabling the Original Caller Information

When the Enable Originating Caller Information box for a trunk group is marked, the function is enabled. To enable the function for forwarding original caller ID:

Step 1 Navigate to Administration–>Trunks–>Trunk Groups.

Step 2 Select an existing trunk group or create a new trunk group.

Step 3 Check the Enable Originating Caller Informatio check box under the Trunk Services heading (Figure 7-21 on page 176). After this enable, the Billing Telephone Number text entry box becomes active and shows the base number of the DID range by default (if DID range has been configured).

Step 4 If necessary, type a value in the Billing Telephone Number field.

Step 5 Click **Save** as needed when other trunk group configuration steps are done.

NOTE Users who should receive forwarded calls must belong to a user group with a class of service (COS) that supports the call forwarding features. Trunk-to-Trunk transfers and Allow External Call Forwarding and Find Me Destinations should be enabled for the COS the user has. For details, see the "Telephony Features Permissions" on page 315.

7.5 Introduction to ISDN Profiles

ISDN profiles let the system administrator specify functions that extend the normal capabilities of ShoreTel trunks. The additional capabilities are enabled in the ISDN profile by the specification of information elements (IEs). After its creation, the purpose-specific ISDN profile is applied as needed to one or more PRI T1, BRI, or PRI E1 trunk groups. ISDN profiles are advanced tools because their purpose is outside the normal use of a ShoreTel Voice Switch.

In the two-stage implementation process, a function-specific ISDN profile with one or more manually typed parameters first is created and subsequently applied to a trunk group.

Example applications of ISDN profiles are as follows:

- For WANs in which a carrier or service provider does not automatically add the CID name, a caller ID (CID) name can be added to outbound calls. For the detailed description of this function, see "Caller ID Name on T1-PRI Trunks" on page 178.
- In Europe, up to 25 digits for a SETUP can be required for ISDN BRI and PRI. To provide these digits, the ShoreTel switch normally passes 20 digits in the SETUP message but can add 5 digits when necessary. For a description of this capability, see "Specifying a 20-Digit SETUP Message for PRI or BRI" on page 185.
- To meet a requirement of compliance testing in Europe, an outbound call can carry the progress indicator value 8. This ISDN profile for this function should be applied on a trunk only during a period of compliance testing.
- In Europe, an ISDN profile can direct the switch to support ISDN channel negotiation by the central office (only for outbound calls from a ShoreTel switch in the current release).

7.6 Caller ID Name on T1-PRI Trunks

The function Caller ID Name on T1-PRI lets a ShoreTel Voice Switch add the user name to caller ID (CID) information in an outbound call. The new function is available on T1-PRI trunks but not BRI or E1-PRI trunks.

Before Release 11.2, ShoreTel Voice Switch did not send the user's name to carriers or service providers. Nevertheless, in the U.S., carriers and service providers could add the caller number and caller name (if the call originates on a ShoreTel Voice Switch).

NOTE Caller ID Name on T1-PRI is optional because most ShoreTel installations utilize the default state of ShoreTel's underlying CID mechanism. Therefore, only certain installations need the functionality of Caller ID Name on T1-PRI. ShoreTel created this function specifically for non-U.S. customers whose carrier or service provider needs the support of Caller ID Name on T1-PRI. Whether this function is really needed can be determined through consultation with the carrier or service provider if the actual need is unclear.

7.6.1 Caller ID Name with the Public Network

This section gives an overview of the current handling of Caller ID names in outbound and inbound calls.

7.6.1.1 Inbound

In North America, some carriers can provide a Caller ID (CID) name in addition to the CID number. In the current release, ShoreTel voice switches support CID name on T1 PRI trunks. A carrier or service provider can deliver the CID name by using either a display message or facility message method. The method depends on the protocol used, as follows:

- In a display message over NI-1
- In a facility message over NI-2

For CID name in an inbound call, ShoreTel switches support both methods at the same time, so no special configuration is required on a ShoreTel switch to accommodate the arrival of CID names.

7.6.1.2 Outbound

ShoreTel T1 voice switches can send a CID name in an outbound call by using either a display message or a facility message. However, in contrast to inbound calls with a CID name, all outbound calls are configured to use either the display message or facility message method. The choice of method for outbound calls must be specified in Director. Even as ShoreTel T1 voice switches support only NI-2 protocol, it is possible to create an ISDN profile for using NI-2 protocol and then specify either the display message or facility message method for outbound CID name delivery. The ISDN profile for this purpose is described in the ISDN Profiles section.

The message method must match the method expected by the carrier or service provider. For example, if a carrier uses NI-1, it might be possible to program NI-2 with the display method such that the carrier accepts the outbound CID name from the ShoreTel Voice Switch.

The steps for selecting the message method and protocol are in the “Implementation” section. The implementation steps are for outbound calls only.

7.6.2 Caller ID Name on T1-PRI Background Details

This section contains additional details that readers should understand before going into the Implementation section.

Outside the U.S.—as in Canada, for example—the carrier or service provider might insert the geographic or metropolitan origin of the call as the CID name (“Coquitlam,” for example). However, to send the actual user name of the caller, some carriers or service providers outside the U.S. require the information provided by Caller ID Name on T1-PRI. Therefore, for customers in Canada and elsewhere who want a CID name to accompany the call, this function enables a ShoreTel Voice Switch to provide the CID name information.

7.6.2.1 Enabling the Outbound Caller ID

Some configuration steps for CID are common regardless of the carrier or service provider. These steps are independent of the Caller ID Name on T1-PRI function but are nevertheless included in the implementation steps. An example of such a mandatory step is the enable for sending CID on an outbound call. This enable is called Enable Outbound Calling Name and is located in the editing window for Switches > Primary. This enable is mandatory for sending a name in the CID, but in some countries a carrier or service provider might require the additional configuration that is supplied by an ISDN profile configured specifically for the purpose of facilitating Caller ID Name on T1-PRI.

7.6.2.2 ISDN Profiles

For Caller ID Name on T1-PRI, a critical element is defined in an ISDN profile. Thereafter, the ISDN profile is applied to a trunk group. ISDN profiles are advanced tools that can be used to specify important information for a variety of features. In the case of Caller ID Name on T1-PRI, the ISDN profile specifies the means for sending the user name (if the carrier or service provider requires this support). The ISDN profile that might be needed for sending a user name is described in the implementation steps.

As has been emphasized, the customer must know what the carrier or service provider expects for the ISDN message method. However, even when the expected method is known, an administrator might have to perform a simple experiment to determine which of the two possible categories of each method is required. The following list shows the possible methods related to CID (see implementation steps for actual use):

CallerIDSendMethod=display

CallerIDSendMethod=displaypcc

CallerIDSendMethod=facility

CallerIDSendMethod=facilitypcc

The choice for using the “pcc” version of a method is not based on information that the carrier provides. For example, if a customer has settled on the display method (CallerIDSendMethod=display) in the ISDN profile and correctly applied the profile to the pertinent trunk group but the CID name is not received at the far end, then an alternative ISDN profile (with CallerIDSendMethod=displaypcc) must be applied to the trunk group. (After an ISDN profile is created, it must also be applied to the appropriate trunk groups, as described in the "Implementing Caller ID Name on T1-PRI" on page 180.)

7.6.2.3 Character Limit and Name Masking

The maximum number of characters that a CID name can support is 34. Although ShoreTel supports up to 34 characters, a carrier or service provider might truncate the CID name. It might, for example, pass only 16 or even 12 characters. This possibility is one of the reasons that customers must know the support provided across the cloud and consult as needed with the carrier or service provider.

NOTE For reasons of privacy, the actual name can be masked by insertion of a generic label, as described in the Implementation section.

7.6.3 Implementing Caller ID Name on T1-PRI

In the configuration descriptions that follow, the first (and mandatory) step for enabling CID name is described. Actually, this step is necessary to enable CID name transmission from any ShoreTel environment. Subsequent sections describe how to specify and apply the ISDN profile with the display method in an environment that requires it for Caller ID Name on T1-PRI.

NOTE Before enabling the Caller ID Name on T1-PRI function, the administrator should have prior knowledge or else consult the customer’s carrier or service provider to determine how it delivers the CID name end-to-end and, therefore, whether the display method is required. (See also Creating and Applying an ISDN Profile for

CID Name.) Furthermore, although the ShoreTel Voice Switch sends the CID name when configured to do so, ShoreTel cannot guarantee the results at the far end. ShoreTel cannot guarantee that carriers or service providers deliver CID names at the far end or that they support overwriting of the user name with a user-specified word (described in the configuration steps). Also, in some cases, parameters on 3rd-party gateways might require modification before the CID name can be delivered.

7.6.3.1 Enabling Outbound Calling Name

The first step for enabling the CID name is to enable the trunk for outbound calling name regardless of whether the network calls for an ISDN profile to activate Caller ID Name on T1-PRI:

Step 1 Navigate to Administration –> Switches –> Primary. The list of primary switches appears. Note the list of switches in the Name column.

Step 2 Click the name of the T1 ShoreTel Voice Switch with the PRI trunk that is to send the CID name. The Edit ShoreGear T1 Switch window opens (Figure 7-22). The switch in this example is Canada PRI T1 - B.

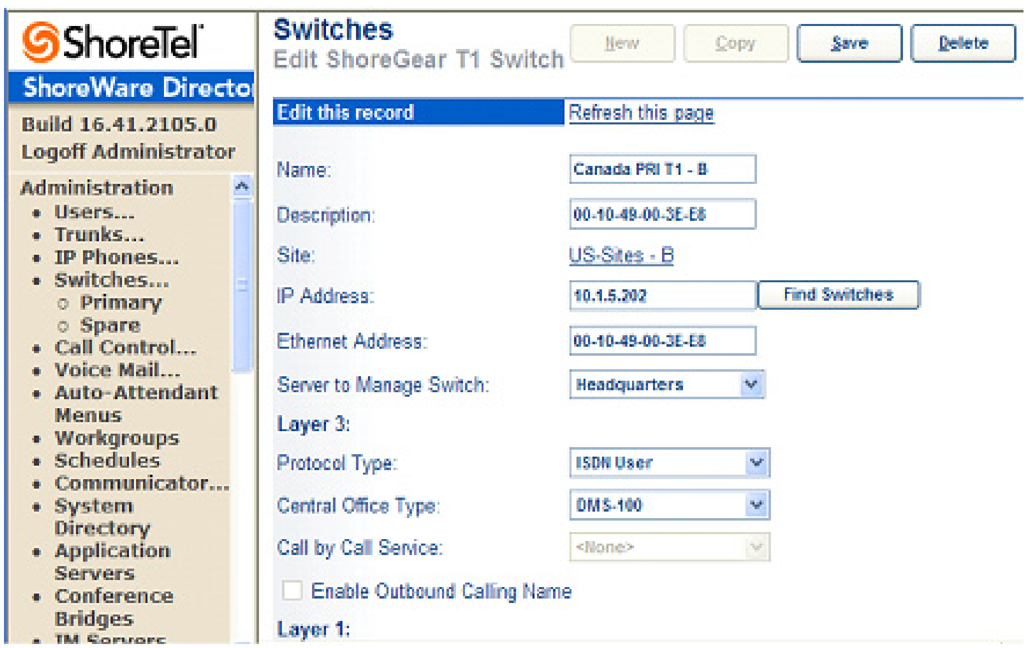


Figure 7-22Enabling Outbound Calling Name

Step 3 Select ISDN User in the Protocol Type drop-down list.

Step 4 Select NI-2 in the Central Office Type drop-down list.

Step 5 Mark the Enable Outbound Calling Name checkbox. This checkbox is just above the area labeled Layer 1.

7.6.4 Creating and Applying an ISDN Profile for CID Name

This section defines the ISDN profile for Caller ID Name on T1-PRI and describes how to apply the ISDN profile to a T1-PRI trunk.

7.6.4.1 Synopsis

In most deployments, the default ISDN system profile is already part of the configuration, and the default ISDN system profile has already been associated with the trunk group. If a new ISDN profile is required, the following tasks are involved and are described in their own subsection:

- Creating an ISDN profile for Caller ID Name on T1-PRI
- Associating the ISDN profile with a trunk group

The ISDN profile for Caller ID Name on T1-PRI specifies a method for delivering a CID name to the carrier or service provider. In all deployments, the default facility method is enabled, but for some WANs, the display, displaypcc, or facilitypss method is required. To use one of these latter message methods, a new ISDN profile must be created.

7.6.4.2 Specifying the Display Method in the ISDN Profile

To meet the interoperability requirements for CID name as needed in some environments, the system administrator creates an ISDN profile that specifies one of two display methods for sending CID names. (If the need for the display method is not a certainty, readers should refer to ISDN Profiles and other conceptual descriptions of the CID Name on T1-PRI function in this section.)

The name for a new ISDN profile must differ from the default profile name SystemISDNTrunk. (Figure 2 shows the default ISDN system profile.)

The system administrator subsequently associates the new profile with a trunk group in the Edit PRI Trunk Group window, as described in the section Enabling Caller ID Name and the ISDN Profile on a Trunk Group.

7.6.4.3 Specifying an ISDN Profile for the Display Method

To specify a new ISDN profile for CID name:

Step 1 Log into ShoreTel Director.

Step 2 Click **Administration > Trunks > ISDN Profiles** to open the ISDN Profiles page shown in Figure 7-23.

Figure 7-23ISDN Profiles Window in Director

Step 3 Click **New** to open Edit ISDN Trunk Profile, as shown in Figure 7-24.

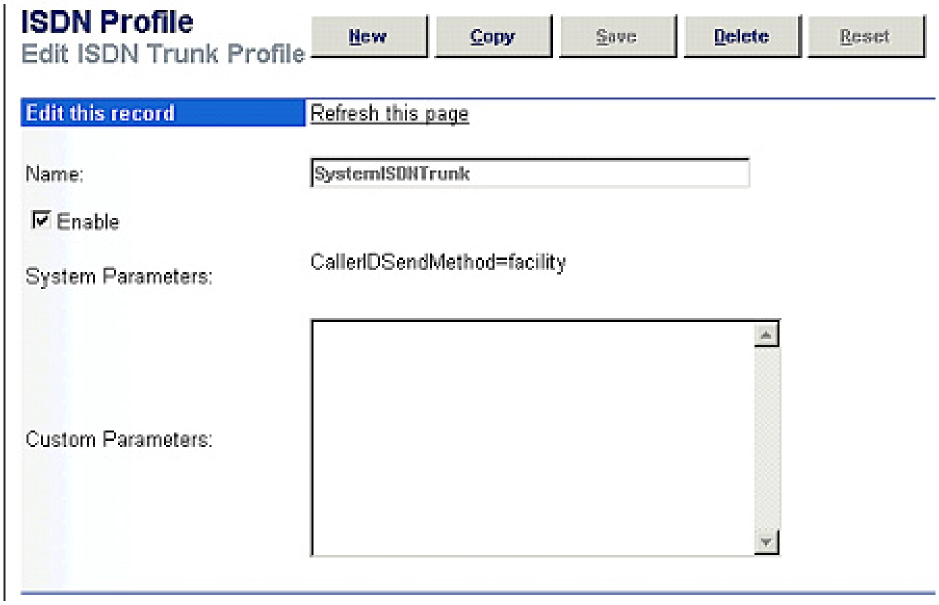


Figure 7-24Default ISDN Profile

Step 4 In the Name field, type the name that you want to use for this ISDN profile. You cannot use the name "SystemISDNTrunk."

Step 5 Check the **Enable** checkbox.

Step 6 In the Custom Parameters area, type the following:

CallerIDSendMethod=display (or **displaypcc**, as needed)

Step 7 Click **Save**.

The next task is to associate the ISDN profile with one or more T1-PRI trunks.

7.6.4.4 Enabling Caller ID Name and the ISDN Profile on a Trunk Group

If an ISDN profile with the facilitypcc, display, or displaypcc method exists, the system administrator associates it with a correct T1-PRI trunk, as described in this section. Not shown here is how the system administrator configures the PRI trunk group in a manner that is similar to other PRI trunk groups (but in this case, an ISDN profile is associated with the trunk).

At the bottom of the Edit PRI Trunk Group window, the administrator enables (or disables) the caller ID function by checking (or unchecking) the Enable Caller ID box.

Many companies might choose to hide the personal name of the caller and instead insert the company name. As the following steps show, a text box associated with CID name lets the system administrator specify a label to overwrite all outbound CID names (and thus mask the call initiator's name). The behavior is as follows:

- If the administrator checks the Enable Caller ID box and leaves the 'Overwrite Caller ID with' field empty, the caller's name goes to the carrier or service provider.

- If the administrator checks the Enable Caller ID box and types text in the 'Overwrite Caller ID with' field, that text is presented to the carrier or service provider.

To configure a PRI trunk group to present the CID to the service provider:

Step 1 Log onto ShoreTel Director.

Step 2 Click **Administration > Trunk Groups**. The Trunk Group pages shown in Figure 7-25.

Figure 7-25Trunk Groups

Step 3 Select a PRI trunk group by clicking on the trunk group name. The Edit PRI Trunk Group page shown in Figure 7-26 opens.

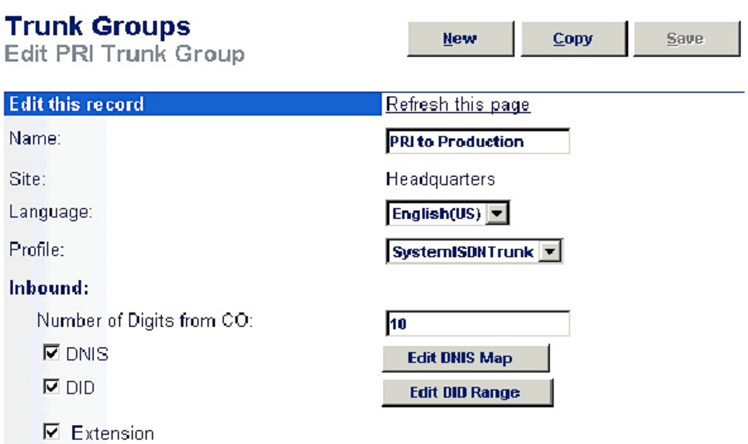


Figure 7-26Selecting the ISDN Profile for CID in the Trunk Group Edit Page

Step 4 Select the appropriate ISDN Profile by using the Profile pull-down menu.

Step 5 At the bottom of the Outbound section of the Edit PRI Trunk Group, put a check in the Enable Caller ID checkbox. See Figure 7-27.

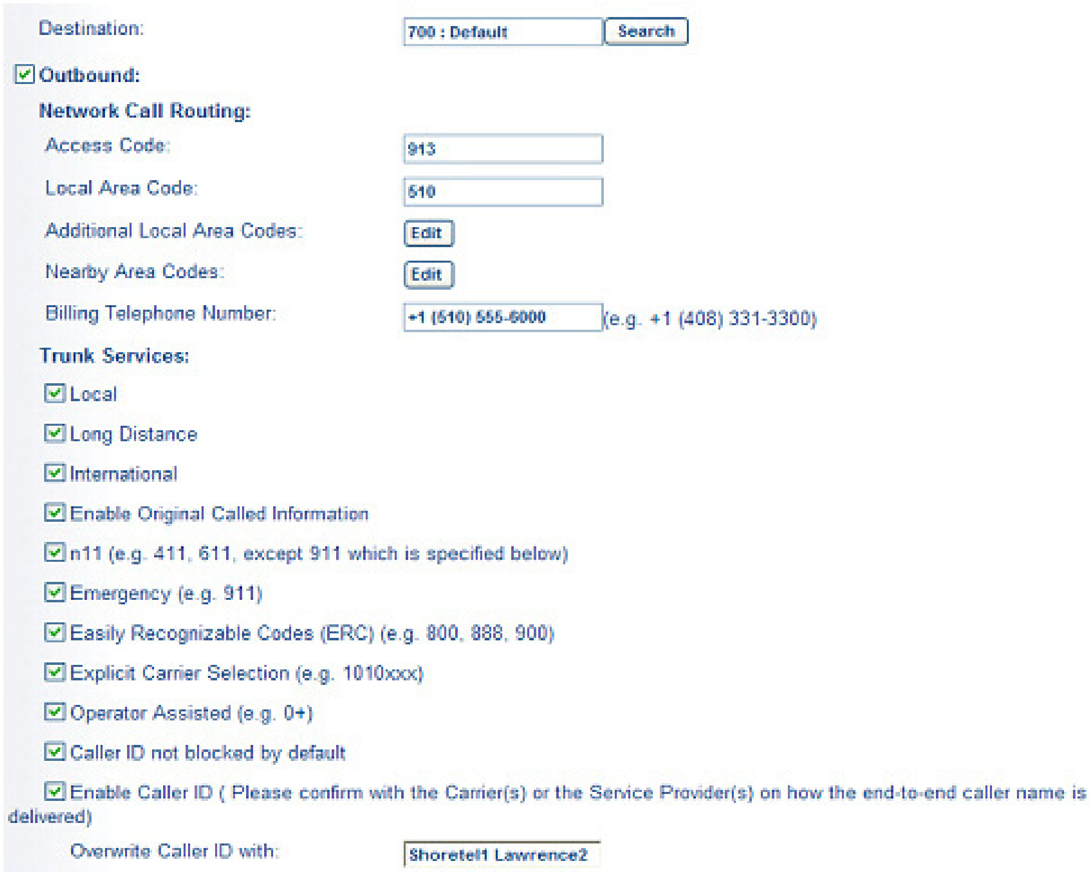


Figure 7-27Enabling the Caller ID Feature

Step 6 (Optional) As needed, type a label for overwriting all individual caller names with this label in the CID display at the far end.

Step 7 Click the **Save** button at the top of the Edit PRI Trunk Group window.

7.7 Specifying a 20-Digit SETUP Message for PRI or BRI

In Europe, a SETUP message with up to 25 digits can be required for ISDN BRI and PRI trunks. To provide 25 digits when necessary, a ShoreTel Voice Switch adds 5 digits to the 20 digits it normally sends in the U.S.

After the terminal equipment (TE) initially receives the SETUP ACK message from the network terminal (NT), the ShoreTel TE can send five digits to the network terminal in the subsequent INFO message if the situation requires those digits.

For the implementation of this messaging, the ShoreTel switch indicates when the extra 5 digits are *not* needed (thus, a 25-digit SETUP message is the default). As the configuration steps illustrate, a message named Sending Complete indicates that the additional 5 digits are *not* needed. Note that, by itself, the Sending Complete message does not directly pertain to the European requirement for 25-digits in a SETUP message. It just says that, for no specific reason, more digits are not required. This message is delivered in 1 of 2 ways:

- The Setup Complete message can go out after the SETUP ACK message arrives. This behavior is the default and does not involve an ISDN profile at all.
- The administrator can specify that Sending Complete goes inside the SETUP message.

To create an ISDN profile in Director that specifies the Sending Complete transmission:

Step 1 Click **Administration > Trunks > ISDN Profiles** in Director. As Figure 7-28 shows, the ISDN Profiles window lists all existing ISDN profiles by name, the enable status of each profile, and a checkbox for selecting a profile.

Figure 7-28List of Existing ISDN Profiles

Step 2 Click the **New** button to start a new profile. The Edit ISDN Trunk Profile window opens, as shown in Figure 7-29.

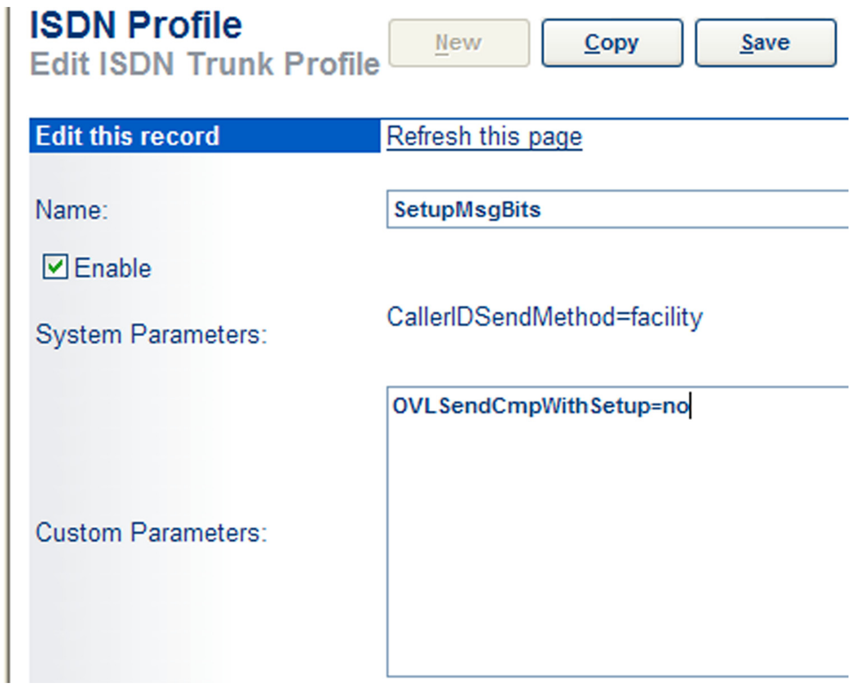


Figure 7-29Creating a New Profile for Sending Complete Message after SETUP ACK

Step 3 In the Name field, type a name for the profile.

NOTE The name of the default profile SystemISDNTrunk is reserved and cannot be used for new profiles. Put another way, the default ISDN profile cannot be modified.

Step 4 In the Custom Parameters area, type one of the following two strings:

OVLSendCmpWithSetup=yes

OVLSendCmpWithSetup=no

where **yes** means the Sending Complete message is carried in the SETUP message, and **no** means the Sending Complete message is transmitted by the ShoreTel Voice Switch after the switch receives the Setup Acknowledge message.

Step 5 Put a mark in the Enable checkbox to enable this profile.

Step 6 Click **Save**.

When ready to apply the ISDN profile to a trunk group, select the profile in the drop-down scroll list in the Trunk Group editing window for the targeted trunk group. After you click the Save button (after completing any other configuration steps), the ISDN profile applies to the trunk group.

7.8 ISDN Profile for RNIE

This section defines the ISDN profile for the Redirecting Number Information Element (RNIE) and describes how to apply it to a trunk group. This profile's purpose is to ensure delivery of an original caller ID to a remote device when the carrier or service provider does not validate the caller ID fields and, therefore, has no strategy for delivering the caller ID. Specifying RNIE ISDN profiles is an advanced task for a network scenario where the default state of the trunk group is not appropriate *after* original caller ID is enabled (as described in "Forwarding Original Caller ID Outside a ShoreTel Network" on page 174).

NOTE No consultation with the carrier is necessary for the use of an RNIE ISDN profile even though, in general, knowledge of how the carrier or service provider communicates at the trunk level is helpful.

7.8.1 Number Sequence in the Q.931 SETUP Message

The RNIE ISDN profile determines which of two possible sequences the ShoreTel Voice Switch places the caller ID and the RNIE in the outbound Q.931 SETUP message. The two sequences are simply the reverse of each other. The correct sequence depends on how the carrier processes the SETUP message. Rather than depending on operational information from the provider, the customer decides on which ISDN profile to use based on whether the original caller IDs are reaching the far-end destination after a call-forwarding feature sends the calls out an ISDN trunk.

After trying both sequences in the ISDN profiles, if the original caller ID does not reach the far-end device, the customer should ensure that the other configuration requirements of caller ID are correct. As described later in this section, even if the ShoreTel Voice Switch is correctly performing its task of forwarding the original caller ID outside the local network, conditions in the WAN might obstruct the successful arrival of caller ID at the destination.

The outbound Q.931 SETUP message has two caller ID fields, as follows:

- One number is the caller ID. This number is either of the following:
  - — The ShoreTel user's DID or the contents of the Billing Telephone Number field when the user has dialed out
  - — The number of the outside caller who called the ShoreTel user and whose call was subsequently forwarded out the trunk
- The other number is the redirecting number—the ShoreTel number that forwarded the original call if call-forwarding was performed. (If the call is not forwarded, there is no redirecting information.) The redirecting number can be one of the following:
  - — The DID of the ShoreTel user who forwarded the call (if the user has a DID).
  - — When necessary, the contents of the Billing Telephone Number: this redirecting number can be the base number in a trunk's DID range, the ShoreTel customer's BTN, or the CESID. These alternatives to the DID are described in "Purpose of the Billing Telephone Number for Caller ID" on page 175.

    NOTE If a carrier forwards an original caller ID to yet another provider, such as a CLEC, the subsequent provider might reject the call. This call rejection could be due to the subsequent provider utilizing the caller ID and RNIE in the opposite sequence of the first provider to transport the call. In this case, when the customer has determined that forwarded calls are being rejected at the far end and has consulted with ShoreTel TAC, an ISDN profile cannot help. The way to re-establish delivery of the forwarded calls through the CLEC—but without original caller ID—is to disable the Enable Original Caller Information until carriers and service providers find a way to interoperate in a way that ensures delivery of caller ID.

NOTE Different carriers or carrier regions can use different call parameters. Therefore, we recommend that a unique ISDN profile be created for each trunk group.

7.8.2 Creating an RNIE ISDN Profile

An RNIE ISDN profile can have one of two lines that specify its effect. The line must be typed according to the following syntax:

SEND_BTN_AS_RNIE=<yes|no>

Default: SEND_BTN_AS_RNIE=yes

The meaning of the Yes/No selection is as follows:

SEND_BTN_AS_RNIE=no: The calling party number is presented with Billing Telephone Number (BTN). The *caller ID* is sent as the redirecting number (if “Enable Original Caller Information” is checked).

SEND_BTN_AS_RNIE=yes: The calling party number is presented with the caller ID. The contents of the Billing Telephone Number field is sent as the redirecting number (if “Enable Original Caller Information” is checked).

To create an RNIE ISDN profile:

Step 1 Click **Administration >Trunks > ISDN Profiles** (or SIP Profiles for the same functionality on SIP trunks).

Step 2 Click **New** at the top of the window (or the name of an existing profile as needed). See Figure 7-30.

Figure 7-30RNIE ISDN Profile

Step 3 Type a name for a new RNIE ISDN profile, such as RNIE2. The name of the default profile (SystemISDNTrunk) cannot be used.

Step 4 Put a check in the Enable checkbox.

Step 5 Do one of the following:

- To specify that the switch presents the contents of the Billing Telephone Number field from the trunk group's configuration, type the following string in the Custom Parameters field:

  SEND_BTN_AS_RNIE=no

- To specify that the switch presents the caller ID, type the following string in the Custom Parameters field:

  SEND_BTN_AS_RNIE=yes

NOTE This setting is also the default state of the trunk group.

Step 6 Click **Save** when each profile is done.

When the time comes to apply the profile to a trunk group, select the profile's name in the Trunk Groups editing window (near the top of the window) for the selected trunk group.

CHAPTER 8

Configuring IP Phones

This chapter describes configuring IP phones. The topics include:

8.1 Overview

ShoreTel supports IP phones connected through ShoreTel Voice Switches. Before you can add IP phone users through ShoreTel Director, you must:

- Set the boot parameters in the individual IP phones if you are using static IP addresses. For more information, see your ***ShoreTel 12: Planning and Installation Guide***.
- Add and configure all ShoreTel Voice Switches that will be supporting IP phones. For information on allocating switch ports to IP phone support, see Chapter 5: Configuring Switches starting on page 89.

For more information on these steps, see the instructions given in the ***ShoreTel 12: Planning and Installation Guide***. When you have completed the installation process, the ShoreTel system automatically detects IP phones connected to the network.

Click the IP Phones link in the navigation frame to expand and display additional links, as shown in Figure 8-1.

8.2 System Settings – IP Phones

8.2.1 Setting IP Address Ranges

If a system includes more than one site (Headquarters), you must define an IP address range for ShoreTel IP Phones at each site in the system. To set ranges for each site ensures that new added phones are associated with the correct site.

ShoreTel Converged Conferencing uses IP phone ports to support conference call ports. Include the capacity of any ShoreTel Converged Conference Bridges in the IP address range calculations.

You can review all the IP phones in the system from the IP Phone List page (see "Viewing IP Phones" on page 202). You can also view IP phones by the switch supporting them from the IP Phone Maintenance page. For more information, see Chapter 19: ChapName on page 537.

To set the IP address range, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > IP Phones > IP Phones Address Map**. The IP Address Map List page appears as shown in Figure 8-1. The page lists the sites and associated IP address ranges.

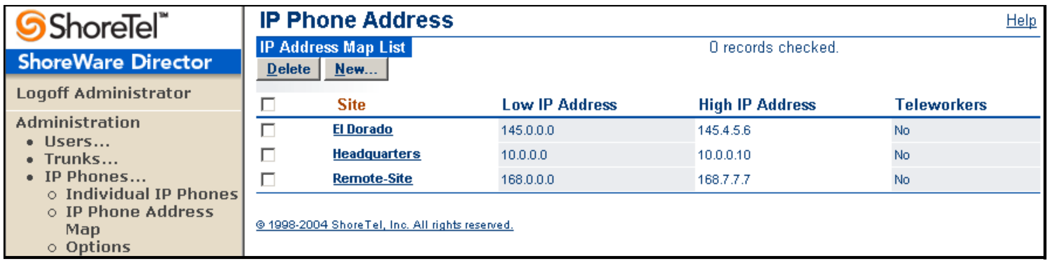


Figure 8-1 IP Address Map List Page

Step 3 Click the **New** button. The Site IP Address Range edit page appears as shown in Figure 8-2.

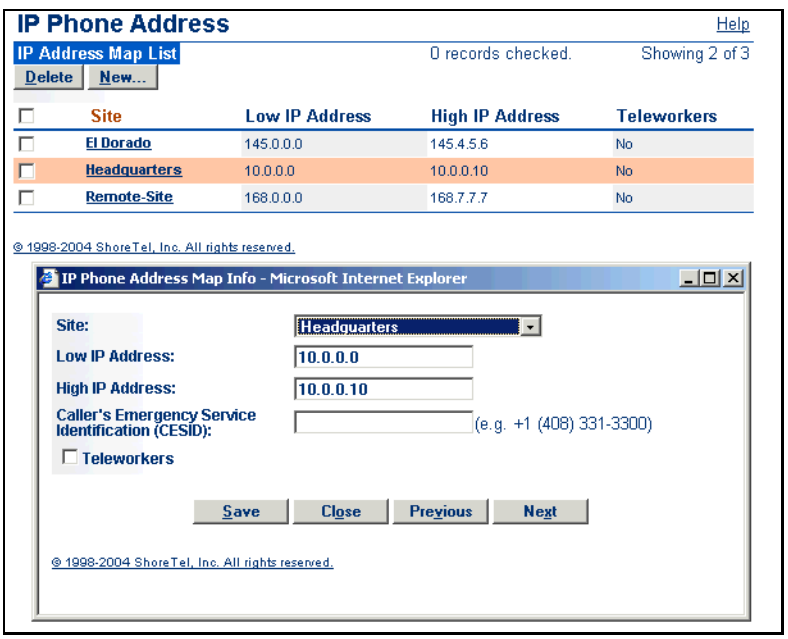


Figure 8-2 Site IP Address Range Edit Page

Step 4 In the Site field, select the site for which you want to configure the IP address range.

Step 5 In the Low IP Address field, enter the lowest IP address that you want to use for the site. The IP Address must be valid for the network in which the site is located.

Step 6 In the High IP Address field, enter the highest IP address that you want to use for the site. The IP address must be valid for the network in which the site is located.

Step 7 In the Caller's Emergency Service Identification (CESID) field, type the Caller ID number that the system passes to emergency responders when an emergency call originates on an IP phone at the site. See A, starting on page595, for information about configuring a system for emergency calls.

Step 8 Check the **Teleworkers** check box if teleworkers will use the site.

Step 9 Click **Save**. The information for the site is listed on the IP Phone Address page.

8.2.1.1 About Teleworking IP Phones

Improved support for teleworking IP phones does not require a dedicated site and ShoreTel switch for the best codec selection and correct bandwidth management. IP phones that fall into the IP address ranges for any teleworking IP phones operate just as other IP phones with a few differences:

- The inter-site voice codec will be used for all calls to and from teleworking IP phones.
- For bandwidth management purposes, the teleworking IP phone is not associated with any site and calls to and from that phone will not deduct from the available bandwidth of the call agent switch for that phone.
- The IP phone will be hosted by an available call agent switch at the site associated with the IP address map. Numbers dialed from teleworking IP phones will be interpreted in the context of the dial plan for the site associated with the IP address map.

8.2.2 Setting IP Phone Options

To set IP phone options such as passwords and configuration switches, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > IP Phones > Options**. The IP Phone Options page appears as shown in Figure 8-3.

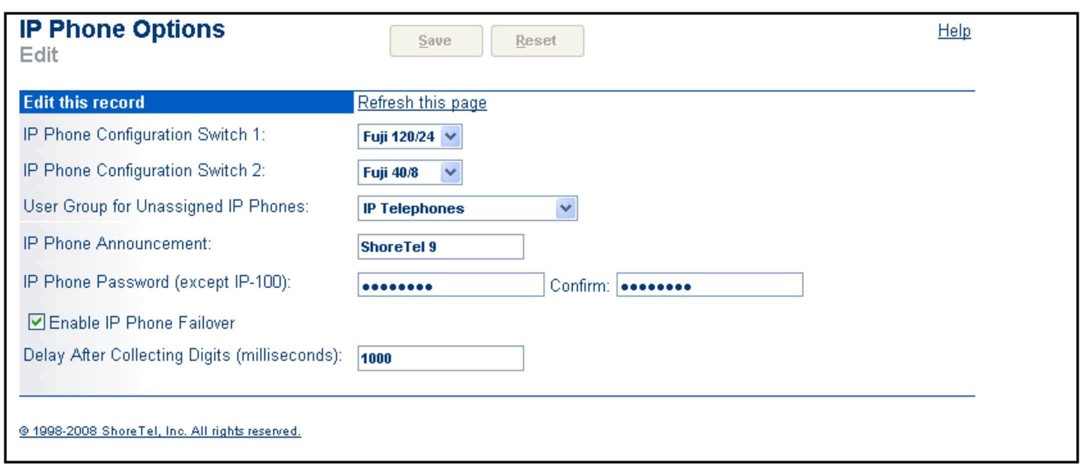


Figure 8-3 IP Phones Options Edit Page

The parameters on the IP Phone Options edit page are defined as follows:

- **IP Phone Configuration Switch 1 and IP Phone Configuration Switch 2:** You need to designate the switch that serves as the phone's Communicator. This switch must be managed by the headquarters server. The IP addresses of these switches are downloaded to the IP phones whenever the IP phones are booted. These switches communicate with the ShoreTel server to determine which switch manages calls for a particular IP phone. You have the option of assigning two switches to this function, in case one switch fails. Every installation with IP phones must have at least one configuration switch.

  To assign configuration switches, simply select an available switch from the drop-down lists for configuration switches 1 and 2. In a new installation, the ShoreTel system automatically assigns the first two ShoreTel-120/24, ShoreTel-60/12, or ShoreTel-40/8 switches you configure as the configuration switches.

- **User Group for Unassigned Phones:** Unassigned IP phones are available for users configured for Any IP Phone. From the drop-down list, select the user group that has the call permissions you want unassigned IP phones to have.

- **IP Phone Announcement:** Enter a message of the day that displays on all IP phones except the ShoreTel IP Phone IP210 phones. The text can be up to 19 characters long and appears left-justified on the phone display. If you wish to center the message, add leading spaces.

- **IP Phone Password:** This field is used only with ShoreTel IP Phones that require a password. It is initially set to "1234". It may be blank or be 1 through 8 digits long. It sets the default password for new IP phones coming online.

- **Enable IP Phone Failover:** When this box is checked, IP phones send a keep-alive message to their Communicator switch every four minutes. If a response is not received, the IP phone attempts to contact an alternate Communicator.

  Changing the state of this field requires a reboot of all IP Phones. The process can take several minutes. Phones in the process of rebooting may drop calls.

  Refer to "Failover – Call Continuation" on page 196 for information about IP Phone Failover

- **Delay After Collecting Digits:** This field represents the timeout period for transferring calls. Instead of having to press a soft key to initiate a call transfer, the desired operation will occur automatically at the expiration of a configurable timeout period. Once all of the necessary digits have been entered (which could vary based on the site's dialing plan), digit collection stops and the timeout period begins counting down. At the end of the countdown, which can be as short as one second, the call is transferred.

  The following features will also be affected by this change:

  — Conference

  — Dialing from the Directory

  — Intercom

  — On-hook dialing

  — Park

  — Pickup

  — Redial

  — Transfer

— UnPark

The timeout period can only be set once for the entire system. You cannot configure different timeout periods for different features or for different users. The timeout period cannot be configured via Communicator or the IP phone interface.

This automatic transfer behavior applies only to blind transfers. If a consultative transfer is placed, the call will remain in the call stack until the far end answers.

8.2.3 LLDP-MED and IEEE 802.1x Support

The Link Layer Discovery Protocol-Media Endpoint Discovery (LLDP-MED) is an enhancement to the Link Layer Discovery Protocol (LLDP) that addresses the following:

- Device location discovery to support location databases and VoIP E911 services.
- Auto-discovery of LAN policies to support to "plug and play" networking.
- Extended and automated power management of PoE endpoints.
- Inventory management

A component of IEEE 802.1 group of networking protocols, 802.1x addresses port-based Network Access Control. It provides an authentication mechanism for devices attaching to a LAN port to either establish a point-to-point connection or prevent the device from accessing the port if authentication fails. 802.1x is based on the Extensible Authentication Protocol (EAP).

8.2.3.1 Implementing LLDP-MED

ShoreTel implements LLDP-MED implementation through the following:

- ShoreTel IP Phone parameter: LldpEnable

  LldpEnable is a ShoreTel IP Phone parameter that consists of one ASCII character. Valid settings for LldpEnable are 0 and 1:

  — LldpEnable = 1 (LLDP is enabled)

  — LldpEnable = 0 (LLDP is disabled)

  Default value is 1 (enabled)

- Modify Setup Source Preferences

  When setting up ShoreTel IP Phones, parameter settings are obtained from the following sources, listed in order of highest to lowest precedence:

  — Config File

  — DHCP (if active)

  — LLDP (if active)

  — Setup Command

  — Default values

Implementing LLDP-MED requires LLDP-MED capable network switches. ShoreTel IP Phones IP110 and IP210 do not support LLDP-MED.

8.2.3.2 Implementing IEEE 802.1x

ShoreTel IP Phones support 802.1x network authentication and enables 802.1x by default. Authentication requires the device to present an ID and password for the user. The default

SID (user ID) is the last 6 characters of the MAC address of the phone, but the password must be manually entered when the phone boots for the first time. The password is cached if authentication succeeds.

If 802.1x enabled on the phone and the network is not set up to handle the feature, the phone boots normally.

Upon an upgrade from another firmware version that supports 802.1x (3.3.x or 3.4.x), the previous settings (802.1x on/off, SID, password) are preserved. Upon an upgrade from another firmware version that does not support 802.1x (2.2, 2.3, 3.1, 3.2), Logical Link Discovery Protocol (LLDP) is turned on by default, and a default SID of the last 6 characters of the MAC address are applied.

While 802.1x is enabled by default in ShoreTel 11 and higher, 802.1x might have been explicitly enabled in earlier releases through the ShoreTel IP Phone parameter 802.1xEnable (a 1-character ASCII parameter). If 802.1x is enabled on the ShoreTel IP Phone and disabled on the network switch, the ShoreTel IP Phone never comes up.

Valid settings for 802.1xEnable are 0 and 1:

- 802.1xEnable = 1 (802.1 authentication is enabled).
- 802.1xEnable = 0 (802.1 authentication is disabled).

  Default value is 0 (disabled).

NOTE ShoreTel IP Phones IP110 and IP210 do not support 802.1x.

8.2.4 Failover – Call Continuation

The IP Phone Calls Continue Through Failover feature maintains active calls through the normal completion of the call even when the Communicator switch handling the call becomes unavailable. IP Phone Calls Continue Through Failover is applicable to ShoreTel IP Phones, SIP extensions, and Conference Bridges. SoftPhone does not support continuation of calls through failover.

8.2.4.1 Failover Description

When a phone's Communicator switch becomes unavailable during a call while IP Phone Failover is enabled, the phone goes through two failover stages:

- Pending Failover is the period between when the phone does not receive the expected acknowledgement signal from its Communicator switch until the time that an alternate switch is assigned to perform Communicator tasks for the phone. This period typically lasts between two to four minutes after the switch becomes unavailable.
- Failover is the period after the alternate switch is assigned to perform Communicator tasks for the phone.

IP Phone Failover is enabled on the IP Phones Options edit page, as described in "Setting IP Phone Options" on page 193.

When a phone enters the pending failover stage, calls in progress remain active. Call control options, including softkey operation, are unavailable during this time. Users cannot initiate new calls and the phone display remains frozen during pending failover.

When IP Phone Calls Continue Through Failover is not enabled, all active calls are dropped when the phone enters the failover stage. The phone then resumes normal operation, allowing the user to initiate or receive calls and perform call control actions on new calls.

When Opaline Calls Continue Through Failover is enabled, active calls are maintained through the beginning of the failover stage until the normal completion of the call. All pending failover restrictions remain in place after the phone enters the failover stage until calls maintained through Failover initiation are completed.

8.2.4.2 IP Phone Behavior during Failover

This section describes IP Phone Calls Continue through Failover behavior on supported devices and references the following switch and endpoint terms:

- **Failed switch**: The switch that became unavailable to trigger the failover process.
- **New switch**: The switch that replaces an unavailable switch in response to a failover.
- **Local endpoint**: An endpoint controlled by the failed switch during a failover.
- **Remote endpoint**: An endpoint controlled by the functioning switch during a failover.

ShoreTel IP Phone – Local Endpoint

During the Pending Failover stage, the telephone user interface (TUI) displays a No Service message until the phone is assigned to a new switch. Call control operations are not available on surviving calls. All inbound calls to the local endpoint are routed to the destination specified by the current Call Handling Mode

During the Failover stage, the TUI displays Failover Mode while surviving calls remain active. Pressing phone keys generates a No Service message on the TUI. Call control operations on surviving calls remain unavailable. All inbound calls to the local endpoint are routed to the destination specified by the current Call Handling Mode. After all surviving calls are concluded, the ShoreTel IP Phone returns to normal operation.

ShoreTel IP Phone – Remote Endpoint

ShoreTel IP Phones on remote endpoints during failover calls can hang up the call, place the call on hold, or retrieve the call from hold. All other softkey operations are unavailable for the duration of the call. The ShoreTel IP Phone continues displaying call information until the of the call. Call control operations on other calls remain available.

SIP Phones

This feature causes no changes to messages displayed by SIP devices. Failover procedures and restrictions are applicable to SIP phones. Call control operations initiated from SIP phones on failover calls are not available.

Trunk Behavior

ShoreTel only releases the trunk after the remote side goes on hook. System cleanup procedures, executed every two hours, release trunks that were left hanging.

8.2.5 Adding Individual IP Phone Users

There are two ways to add IP phone users to the system. The first method requires you to assign IP phones to each user, while the second uses the Any IP Phone feature to allow users to assign their own phone from their desktop and voice mail. Using the Any IP Phone method simplifies the setup of multiple new users. You can use either or both methods, depending on your specific installation.

8.2.5.1 Adding Users with Any IP Phone

This section describes how to configure to use an IP phone. For information about creating users, see “Configuring Users” on page -313. To add an IP phone to a user profile:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Users > Individual Users**. The Individual Users page appears.

Step 3 In the “Add new user at site” field, select the site where you want to add a new IP phone user.

Step 4 Click **Go**. The Edit User page appears as shown in Figure 8-4.

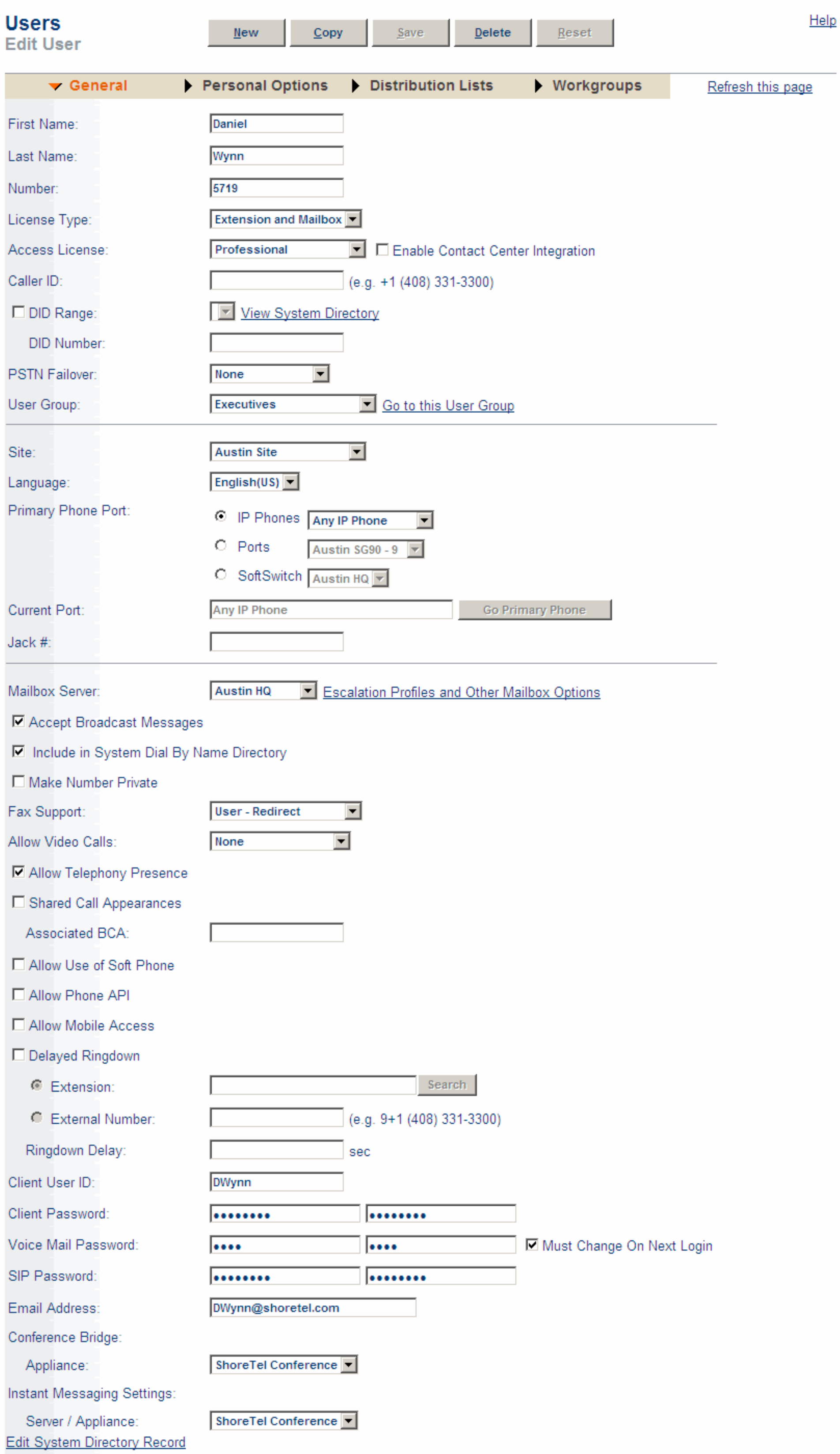


Figure 8-4 Edit User Page

Step 5 Scroll to the Primary Phone Port parameter and do the following:

- Click the **IP Phones** radio button.
- In the field, select the IP phone profile that you want this user to use.
- Click **Save**.

Step 6 Click **Administration > Users > User Group**. The User Groups page appears.

Step 7 Select a user group with a Class of Service telephony profile that allows extension reassignment. For more information about extension reassignments, see "Telephony Features Permissions" on page 315.

Step 8 Click **Save**.

NOTE You can use this profile to create other users. To do so, click Copy and repeat Step 5 through Step 8.

Instruct the users to log in to their voice mail from their desktop and follow the prompts. Each user's extension will automatically be assigned to the IP phone he or she is using. Make sure that users know which phone they should initially use so that phones are not accidently assigned to the wrong users.

All IP Phones are assigned to the "Headquarters" site. When you assign a specific IP phone, the user belongs to the site where the IP phone is located. Only IP phones associated with the currently selected site appear in the IP phone drop-down list.

After users have logged into voice mail to assign their IP phones, you can view the IP phones and users from the IP Phone list page or the IP Phone Maintenance page. For information about editing user information, see "Configuring Users" on page -313.

8.2.5.2 Assigning IP Phones by User

You can also assign IP phones through the Edit User page, one user at a time. You can assign phones by specific IP addresses or assign users to Any IP Phone according to the user's needs.

To assign an IP phone:

Step 1 Click **Users > Individual Users** from the navigation frame.

Step 2 In the Add New User at Site field, select the site for the new user.

Step 3 Click **Go**. The Edit User page appears as shown in Figure 8-4 on page 199.

Step 4 In the Primary Phone Port section, click **IP Phones** and select the IP phone by IP address or select Any IP Phone. Only the IP phones in the current site are in the list.

Step 5 Complete the user profile as required. For information about user settings, see "Configuring Users" on page -313.

Step 6 Click **Save**.

8.2.6 Anonymous Telephones

Anonymous telephones provide flexibility to the ShoreTel system by making additional ports or IP phones available without assigning them to any particular user extension. When configured as anonymous telephones, these ports and IP phones cannot receive calls but do have access to dial tone. If they have the proper Class of Service (CoS) permissions, users can assign their extensions to these phones via the telephone user interface. Once the port or IP phone is assigned an extension, it will receive calls to that extension until the user unassigns it. For more information on how to use the Extension Assignment feature, see the "Using Extension Assignment" on page 410.

The Anonymous Telephones page (Figure 8-5) lets you configure anonymous telephone ports and IP phones. Click the Anonymous Telephones edit page from under the expanded Users link in the navigation frame.

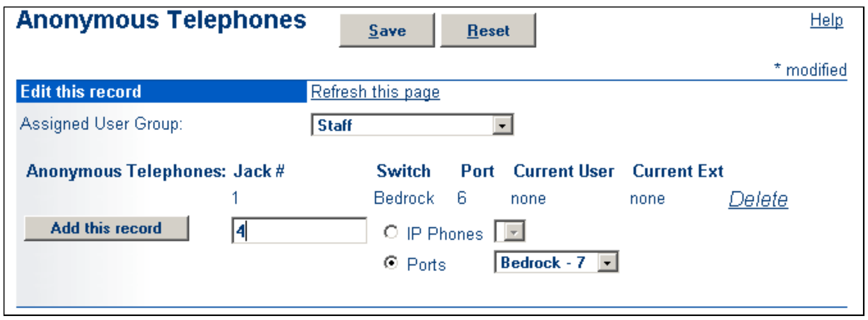


Figure 8-5 Anonymous Telephones Edit Page

The Anonymous Telephones page first lists any Vacated Telephones. A vacated telephone is a telephone that is configured as the home port or IP phone of a user on the system, but that user is currently assigned to another telephone and no other user is assigned to the vacated home phone.

Click Add This Record to add a new anonymous telephone port. Click Delete next to a record to delete an anonymous telephone port from the ShoreTel system. This also disconnects any calls that are in progress on the port.

You can make multiple changes on the Anonymous Telephones page. You must click Save to save the changes.

- **Assigned User Group:** This lets you select a user group that you assign to an anonymous telephone port.
- **Jack #:** This is the name of the telephone jack associated with the vacated port. This is typically the physical telephone jack that the telephone plugs into.
- **Switch:** This is the switch that the vacated telephone port is associated with. It can be either an analog port or an IP phone.
- **Port**: This is the physical switch port number or IP phone MAC address that identifies the vacated telephone.
- **Current User:** This is the name of the user currently using the anonymous telephone port.
- **Current Ext:** This is the extension of the user currently using the anonymous telephone port or IP phone.

8.3 Viewing and Editing IP Phones on the System

To help you manage the IP phones, ShoreTel Director allows you to view and edit all IP phones on the system. Figure 8-6 shows the IP Phones list and edit pages.

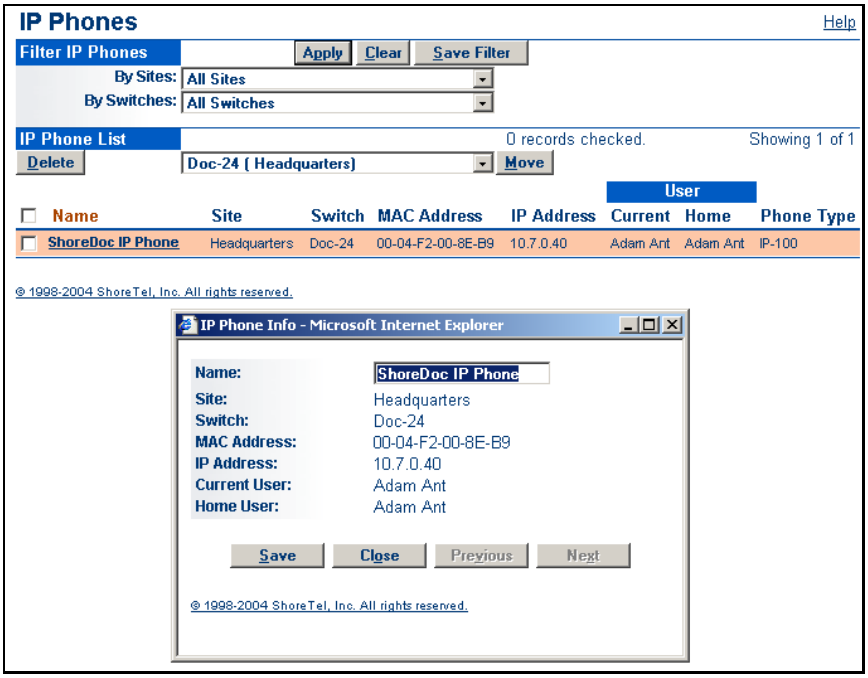


Figure 8-6 IP Phones List and Edit Pages

8.3.1 Viewing IP Phones

To view the IP phones on the system:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > IP Phones > Individual IP Phones** from the navigation frame.

Step 3 In the By Sites field, select the site that you want to view.

Step 4 In the By Switch field, select the switch you want to view from drop-down list.

Step 5 Click **Find Now**. The IP phones for the sites and switches you have selected are displayed.

IP phones can also be viewed from the maintenance pages of ShoreTel Director. For more information, see "Maintenance" on page -537.

8.3.2 Renaming IP Phones

You can change the name of an IP phone from the IP Phone List. By default, IP phones are listed by MAC address in the IP Phone List Name column.

To change the name of an IP phone:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > IP Phones > Individual IP Phones**.

Step 3 In the By Sites field, select the site where the phone for which you want to change the name is located.

Step 4 In the By Switches field, select the switch that the phone uses.

Step 5 Click **Find Now**. The IP phones that satisfy the parameters appear on the page.

Step 6 In the Name column, click the IP phone you want to re-name. The IP Info dialog box appears.

Step 7 In the Name field, enter the name you want to use for the phone.

Step 8 Click **Save**.

8.3.3 Deleting an IP Phone

IP phones can be deleted from the system.

To delete an IP phone:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > IP Phones > Individual IP Phones**.

Step 3 In the By Sites field, select the site where the phone for which you want to change the name is located.

Step 4 In the By Switches field, select the switch that the phone uses.

Step 5 Click **Find Now**. The IP phones that satisfy the parameters appear on the page.

Step 6 Click the check box of the IP phone that you want to delete.

WARNING Make sure that you have the selected the correct phone and that no other phones are selected.

Step 7 Click Delete. A dialog box requests confirmation.

Step 8 Click Yes to delete the phone.

If you wish to add the IP phone back into the system, you must reboot the IP phone. It will be reconfigured during the boot process and become available again. For more information on IP phone configuration, see the ***ShoreTel 12: Planning and Installation Guide***.

8.3.4 Moving an IP Phone

To move an IP phone to a destination switch on a remote site, the remote site must have an IP address range defined. You may not move an IP phone to a switch on a remote site if the IP address of the phone is not within the IP address range of the destination site.

You can move an IP phone across switches at Headquarters without entering an IP address range for the Headquarters site. The IP address range restrictions are only for switches at remote sites.

To move an IP phone:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > IP Phones > Individual IP Phones**.

Step 3 In the By Sites field, select the site where the phone for which you want to change the name is located.

Step 4 In the By Switches field, select the switch that the phone uses.

Step 5 Click **Find Now**. The IP phones that satisfy the parameters appear on the page.

Step 6 Click the check box of the IP phone you want to move.

Step 7 In the Move field, select the switch to which you want to move the phone.

Step 8 Click **Move**. A dialog box requests confirmation.

Step 9 Click **Yes** to move the phone.

8.3.5 IP Phone State Display

The following list explains the states displayed by the IP phones:

- **Available:** Phone has no user assigned to it. Calls can be placed from the phone but it does not receive calls. The Caller ID is "Anonymous".
- **<User Name> <User Ext>:** The phone is assigned to <User Name>.
- **Anonymous:** The assigned user has used the Extension Assignment feature. Calls can be placed from the phone but it does not receive calls. The Caller ID is "Anonymous."

  Another method by which it can become anonymous is if the administrator explicitly configures anonymous phones which do not have assigned users.
- **Unavailable:** The phone was once in the ShoreTel system but has been removed using Director. The phone has no dial tone and is not functional.

8.3.6 Displaying ShoreTel IP Phone Settings

You can display the phone's current IP parameters setting by entering a key sequence from the phone's keypad.

To display the phone's IP parameter settings:

Step 1 Press the **MUTE** key followed by **4636#** (INFO#). The phone will display the first two parameters.

Step 2 Press **#** to advance the display or * to exit. The phone will resume normal operation after the last parameter has been displayed.

8.3.7 Resetting the ShoreTel IP Phone

Phones are reset by a key sequence on the phone's keypad. To reset the phone, a user presses the Mute key followed by 73738# (RESET#). The phone reboots.

8.4 Feature Settings – IP Phones

IP phones can be configured by individual phones types or by User Group settings. The following sections describe how to configure ring tones and wallpaper (displayed graphics) for all phones of the same type. This section describes how to configure ringtones, wallpaper, and application settings on appropriate phones assoicated with specific user groups.

8.4.1 Custom Ring Tones

ShoreTel IP phones offer four different sets of ring tones, with each set consisting of one tone for internal calls and another for external calls. The user is free to select from the four tones as desired. However, in some densely populated work environments, four tones may not be enough for users to be able to distinguish the sound of their phone from that of their neighbors' phones.

To help reduce confusion, most ShoreTel IP phones support the ability to load custom ring tones on an IP phone so that each user can have a unique ring tone.

One set consisting of two custom ring tones can be loaded onto each IP phone. This new set of tones displaces one of the existing sets of ShoreTel ring tones, as shown in Figure 8-7.

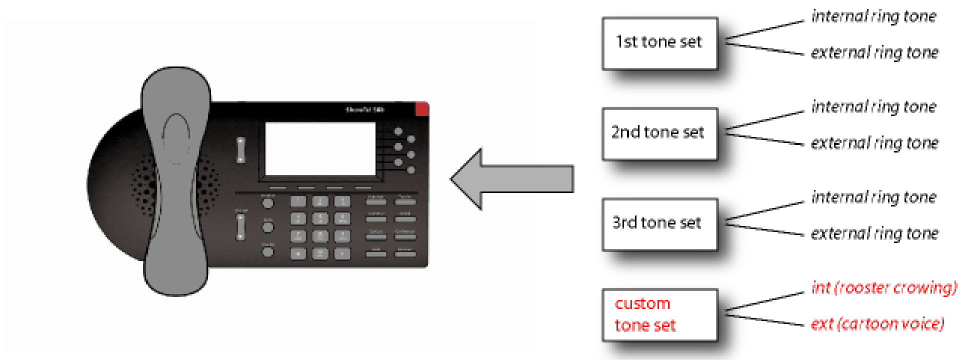


Figure 8-7 The 4th Set of ShoreTel ring Tones Is Replaced by the New Custom Tones

8.4.2 Loading Custom Ring Tones

Ring tones must be in waveform audio file format (WAV). ShoreTel does not offer custom ring tones, but numerous web sites offer free WAV downloads. After a set of custom ring tones has been selected, the system administrator downloads the tones to an IP phone by way of an FTP server. A list of relevant details for custom ringtones follows, and configuration steps follow this list.

- This feature is supported on all ShoreTel IP phones but is not supported on the Polycom IP100 phones.
- ShoreTel offers no tools for creating or managing the custom WAV files.
- Custom files must be loaded in WAV format only.
- The following formats are supported by the phone:
  - — µ-law: 8-bit, 8 kHz, 16 kHz, monaural
  - — α-law: 8-bit, 8 kHz, 16 kHz, monaural
  - — 16-bit, 8 kHz, monaural -or- 16-bit, 16 kHz, monaural
- Two custom tones can be loaded on a phone. Their combined size must be less than 750 KB.
- If a WAV file lasts less than six seconds, the phone loops the ring out to a six-second length before it repeats WAV file.

To load a ring tone onto an IP phone:

Step 1 Identify the WAV files to use (by online search or by creating them manually). The files must reside on a server that is accessible to the IP phone by anonymous FTP. (This server does not have to be the same as the host of the configuration files.)

Step 2 Manually create a phone-specific configuration text file and store it in the same directory as the standard IP phone configuration files. The name of the phone-specific file should contain the MAC address of the target phone for the ring tone. The MAC address can be found on the sticker on the back of the phone. (See the "Creating the Phone-Specific Configuration File" on page 206 for details.)

Alternatively, you can load the same pair of custom ring tones onto other IP phones at the same time, but all of the phones must be the same model (for example, all IP560g phones or all IP212k phones). This approach could cause ring tone confusion if the phones were concentrated in one area of a building.

Step 3 Reboot the phone so that it retrieves the information in the text file and the pointer to the WAV. At boot time, the phone indicates the success or failure of phone-specific configuration download and the WAV download.

8.4.2.1 Creating the Phone-Specific Configuration File

To create a phone-specific configuration file, two new configuration parameters, WaveRinger1 and WaveRinger2 can be inserted into a phone-specific configuration file. These parameters identify the name and location of the custom ring tones that the IP phone will download (via FTP) into the phone's RAM at boot time as shown in Table 8-1.

Table 8-1 WaveRinger1 and WaveRinger2 configuration parameters

WaveRinger1 WaveRinger2	Up to 64 ASCII Characters	Used to assign one Wave File to any of the ring tones defined in Table 2. The first value is the ring tone, and the second value is the location of the file on the FTP server. *Examples:* *WaveRinger1 L/rg 192.168.0.20/audio/dave.wav* *WaveRinger2 L/r1 192.168.0.20/audio/dave.wav*

The name of the phone-specific configuration file should be as follows:

shore_*aabbccddeeff*.txt

where "*aabbccddeeff*" is the MAC address.

To load the same custom ring tone onto several IP phones at the same time, do not use the MAC address in the name of the file. Instead, use the custom configuration file for the phone type being deployed (e.g. S6 for an IP560). For example, the configuration file name for an IP560 phone would be: S6custom.txt

For example, to load one of the custom ring tones, you could replace **L/r14** (i.e. Ring 4 External) and **L/r15** (i.e. Ring 4 Internal) with the name and location of the file containing the new custom ring tone as shown in Table 8-2.

Also note that it is possible to replace internal and external ring tones in separate sets (e.g. Ring 2 external and Ring 4 internal), but only one set of ring tones can be active at a time

Table 8-2 Ring Tones and Symbols

Ring tone	Symbol
Standard - External ring	L/rg
Standard - Internal ring	L/r1
Ring 2 - External ring	L/r10
Ring 2 - Internal ring	L/r11
Ring 3 - External ring	L/r12
Ring 3 - Internal ring	L/r13
Ring 4 - External ring	L/r14
Ring 4 - Internal ring	L/r15

so activating either set of ring tones will only activate only one of the custom rings tones at a time.

8.4.3 Customizing Color LCD Displays with Wallpaper

The ShoreTel 265 and 565g IP phone models offer color TFT-LCD color displays. One of the benefits of having a color display is the ability to download images of your choice onto the phone from a server and display the image as wallpaper.

8.4.3.1 Wallpaper File Specifications

The wallpaper is 320 by 240 pixels and uses an uncompressed 256-color.bmp file format. Each of the 256 colors is defined by a 24-bit RGB value. Bitmap files can be composed using MS Paint or any other editor that can create Paint-compatible files.

To create a graphic file that can be used as a wallpaper image:

Step 1 Open the image in Microsoft Paint.

Step 2 Click **Image > Attributes**. The Attributes dialog box appears.

Step 3 Verify the following parameters are set as follows and make the appropriate adjustments:

- Width = 320
- Height = 240
- Units = Pixels
- Colors = Colors

Step 4 Click **OK** to close the dialog box.

Step 5 Click **File > Save As**. The Save As dialog box appears.

Step 6 In the “Save as type” field, select **24-bit Bitmap (*.bmp;*.dib)**.

Step 7 In the File Name field, enter the name that you want to use for the file.

Step 8 Click **Save**.

8.4.3.2 Downloading the File to Several Color-Screen IP Phones

The wallpaper file that each IP phone displays is specified in configuration files located in the phone configuration directory on the Headquarters server. On standard ShoreTel installations, the phone configuration directory is C:\Inetpub\ftproot.

ShoreTel specifies one text file for each phone model that defines default characteristics for all phones of that model type on the system. You specify the default Wallpaper for phones of a specific model by adding a line to the corresponding configuration file.

To load a wallpaper image onto all system phones of an specific model:

Step 1 Save the wallpaper file on the headquarters (HQ) server in following directory: **C:/Inetpub/ftproot**

Step 2 Access the **C:/Inetpub/ftproot** directory on the HQ server.

Step 3 Open the customization file for the desired phone model:

- For IP 265, open **s36custom.txt**
- For IP 565g, open **s6ccustom.txt**

Step 4 Add the following line to the open file: **Wallpaper2pixmap *abc.bmp***, entering the name of the wallpaper file in place of *abc.bmp*, then save and close the file.

For example, if the wallpaper file is name logo.bmp, then enter Wallpaper2pixmap logo.bmp.

Step 5 Open the configuration file for the desired phone model:

- IP 265: open **shore_s36.txt**
- IP 565g: open **shore_s6c.txt**

Step 6 Verify that the file contains the one of the following lines

- IP 265: **Include s36custom.txt.**
- IP 565g: Include **s6ccustom.txt**.

Add this line to the file if it is not present.

Step 7 Reset the phones.

Step 8 Verify that, when each phone reboots, they download and save the wallpaper file.

Step 9 Verify that each phone displays the new wallpaper file.

8.4.3.3 Downloading the File to a Single Color-Screen IP Phone

Individual configuration files override model default settings for individual IP phones. You can assign custom wallpaper files for individual IP phones by modifying the corresponding configuration file.

To load a wallpaper image onto a single phone:

Step 1 Save the wallpaper file on the HQ Server computer in following directory: **C:/Inetpub/ftproot.**

Step 2 Access the **C:/Inetpub/ftproot** directory on the HQ Server.

Step 3 Create a text file named shore_xxxxxx.txt, where xxxxxx is the MAC address of the phone. Use lower case text when naming the file.

The MAC address is a twelve digit number that uniquely identifies each individual device. This address is printed on the white bar code located on the bottom of the phone.

Example: If the MAC address of an IP 565g is 00104907020C, then create a file named shore_00104907020c.txt

Step 4 Add a line in the open file with the following format: **Wallpaper2pixmap** ***abc*.bmp**, where ***abc*.bmp** is the name of the wallpaper file, then save and close the file. For example, if the wallpaper filename is **logo.bmp**, enter **Wallpaper2pixmap logo.bmp**.

Step 5 Reset the phone.

Step 6 Verify, while the phone reboots, that it downloads and saves the wallpaper file.

Step 7 Verify that the phone displays the wallpaper file

8.4.4 Setting Custom Phone Options for User Groups

ShoreTel 12 and later provide enhanced configuration options for configuring ringtones, wallpaper, and applications on appropriate phones used by User Groups. These phone settings allow phones like the ShoreTel IP 655 to be customized for given users.

Custom phone configurations for groups can be modified from the Users > User Groups menu in Director (see Figure 8-8).

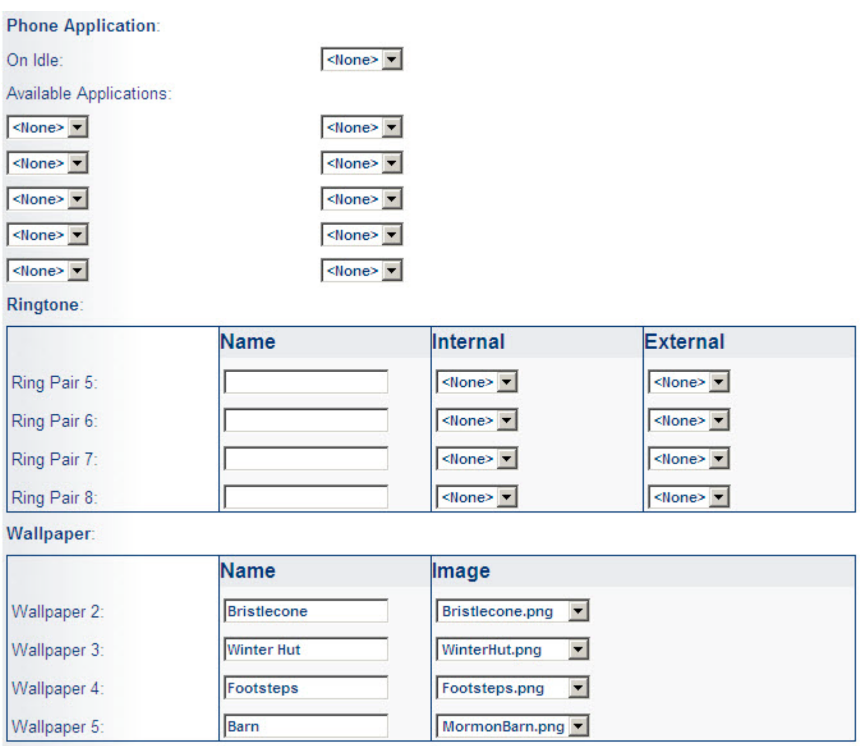


Figure 8-8 Phone customization section of User Groups Edit menu

To configure custom features for user groups, go to the Administration -> Users -> User Groups menu and select the desired user group for the options (or create a new group if desired).

The bottom third of the menu provides fields to configure applications for phones that can use them, such as the ShoreTel IP 655. These applications are referenced by entering a URL that invokes the appropriate application. Additionally, both the ringtones and wallpaper utilized by the phones can be configured from the appropriate fields on this menu.

8.4.5 Wireless Headset Hook Switch

A handset lifter is a device that can be used with phones that do not support the ability to place a call while the phone's handset is on-hook. The device is used in conjunction with a wireless headset worn by the user. When the user wishes to answer a call, she presses an activation button on the wireless headset which signals the handset-lifting device to manually lift the handset, thus generating dial tone and allowing the call to be placed. The small device attaches to the side of the phone and physically lifts the phone handset off-hook and keeps it there for the duration of the call.

As technology has advanced, many phone systems have eliminated the need for a physical handset lifter and have incorporated an electronic version of this device (i.e. electronic hook switch) into the phone design. Thus, the phone can be instructed to go off-hook electronically (as opposed to mechanically) in order to produce dial tone.

ShoreTel has incorporated the electronic handset-lifting functionality into various IP phone models (see below for a list of supported models). These ShoreTel IP phones work with the **Plantronics CS50** wireless headset. Users who have purchased this supported headset model can answer or end calls by pressing the activation button on their headset when they hear their phone ring. If they are too far from their phone to hear it ring, the headset will generate an audible cue to announce incoming calls.

Details:

- This feature is currently supported on the following ShoreTel IP phone models:
  - IP565g
  - IP560g
  - IP560 – Newer models support this feature while older IP560 models do not. (To see if your IP560 supports this feature, flip the phone and check the label with the model number. If the model number ends with a suffix of "-03" or higher, the phone will support this feature. If the suffix ends in "-01" or "-02" the feature is not supported.)
  - IP265
  - IP230
  - IP212k
- Using this feature will defeat auto on-hook and off-hook behaviors.
- The feature is only supported in conjunction with **Plantronics CS50** headsets.

To configure this feature in ShoreTel Director:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Users > Individual Users**. The Individual Users page appears.

Step 3 Select the user for whom you want to set this feature. The Edit User page appears.

Step 4 Click the Personal Options tab.

Step 5 Scroll down to Automatic Off-Hook Preference and click the **Wireless Headset** radio button.

Step 6 Click **Save**.

To configure this feature on an IP phone via the telephone interface:

Step 1 Press the **Options** button on the IP phone, followed by the associated password and the # key.

Step 2 Scroll to option **2: Auto Off-Hook**.

Step 3 Press the Edit soft key to select this option.

Step 4 Scroll to option **3: Wireless Headset**.

Step 5 Press the **OK** soft key to confirm.

Step 6 Press the **Done** soft key to confirm.

8.4.6 Programmable IP Phone Buttons

An administrator or user can change the functions associated with the ShoreTel IP Phone or B "Programmable IP Phone Buttons" on page 211 custom buttons. Users can create shortcuts for operations that would normally require pressing two or three buttons to accomplish the same task.

The action associated with the bottom button on an IP 530/560 could be configured to speed dial a particular extension or external number. The button above that could be set to perform overhead paging, and so on. All of the custom buttons are configurable except for the top-most button, which is permanently set to provide call appearance information (i.e. ringing indicator and call timer information).

Users can keep track of which actions they perform on a regular basis and then associate those actions with the custom buttons. Instead of having to dial a star code (such as *14 for picking up the Night Bell) they can just press one button. See Table 8-3 for supported functions.

Table 8-3 Supported Programmable Button Functions:

Function	Parameter	Comments
Agent Login	None	
Agent Logout	None	
Agent Wrap-Up	None	
Barge In	Extension or none	
Bridged Call Appearance	Extension or none	
Call Appearance	None	Not supported on Button Box
Centrex Flash	None	Refer to Section 7.2.11 on page 170.
Change CHM	Change Call Handling Monitor	

Table 8-3 Supported Programmable Button Functions:

Function	Parameter	Comments
Change Default Audio Path	Audio Call Path	
Conference Blind	Extension or external number	
Conference Consultative	Extension or external number	
Conference Intercom	Extension or none	
Dial Mailbox	Extension or none	
Dial Number (Speed Dial)	Extension or external number	
Group Pickup	Extension	
Hotline	Extension	
Intercom	Extension or none	
Monitor Extension	Extension or none	
Page	None	
Park	Extension or none	
Park and Page	Extension or none	
Pickup	Extension or none	
Pickup Night Bell	None	
Pickup/Unpark	Extension or none	Uses internal presence to determine which operation to perform
Record Call	Mailbox	Operates on selected call in connected; Pressing a second time stops the recording. Call recordings can be saved in the mailbox of the initiating client (by leaving the mailbox field blank), or can be routed to an alternate mailbox by entering a mailbox number in the field.
Record Extension	Extension or mailbox	Call recordings can be saved in the mailbox of the initiating client (by leaving the mailbox field blank) or can be routed to an alternate mailbox by entering a mailbox number in the field.
Send Digits Over Call	Extension	
Silent Coach	Extension	
Silent Monitor	Extension or none	
Toggle Handsfree	None	

Table 8-3 Supported Programmable Button Functions:

Function	Parameter	Comments
Transfer Blind	Extension or external number	
Transfer Consultative	Extension or external number	
Transfer Intercom	Extension or none	
Transfer to Mailbox	Extension or none	
Transfer Whisper	Extension or none	
Unpark	Extension or none	
Unused	None	
Whisper Page	Extension or none	
Whisper Page Mute	None	

Details:

- After a function is assigned to a button, users can enter a label (up to five characters on multiline ShoreTel IP Phones and up to six characters on the IP 100 and BB24). The label appears on the LED display next to the custom button.
- The system administrator can configure the custom buttons via ShoreTel Director on behalf of a user, or he can enable permissions for an individual user so that the user can modify the custom buttons on his own IP phone via the telephone interface.
- The programmable button feature is supported on the multiline models but is not supported on the IP 210.

8.4.6.1 Configuring Programmable Buttons via Director

To configure programmable buttons via ShoreTel Director, follow the procedure below:

Step 1 Launch ShoreTel Director.

Step 2 Click on the **Administration > Users > Individual Users**.

Step 3 Click the name of the user whose phone you want to modify.

Step 4 Click the **Personal Options** tab.

Step 5 Click the **Program IP Phone Buttons** link. The Program IP Phone Buttons page appears as one shown in Figure 8-9.

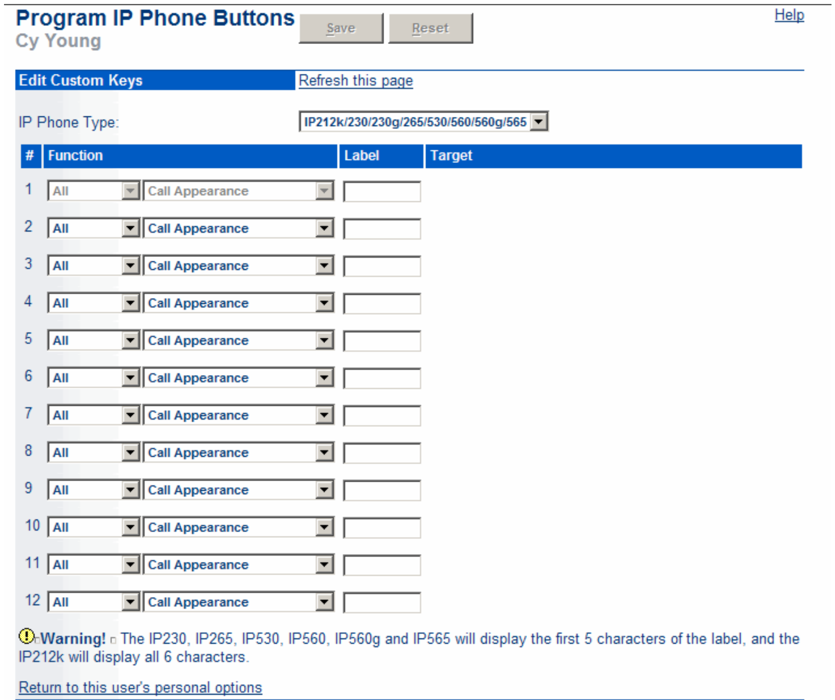


Figure 8-9 Program IP Phone Buttons Page

Step 6 In the IP Phone Type field, select the type of phone the user is using.

Step 7 For each button # that you want to configure, do the following:

- Step a In the first Function field, select the category that you want to use for this button.
- Step b In the second field, select the action you would like to associate with a particular button. For information about the actions, see Table 8-3 on page 211.
- Step c In the Label field, enter a descriptive word to remind the user which function is associated with that button. This label can be up to five characters long for the multiline phones and up to six characters long for the IP 100 and BB24.The label will appear on the LED display adjacent to the button.
- Step d When applicable, enter the appropriate information in the fields that appear in the Target section.

  Certain functions do not require entering a destination, but other functions such as speed-dial or blind transfer can optionally take a destination. Some functions take only extensions and some take any type of phone number.

Step 8 Click **Save**.

8.4.6.2 Configuring a Hotline Button

A Hotline is a bi-directional ringdown circuit accessed through IP Phone or Communicator buttons. A hotline call is initiated by pressing the assigned button. Hotline calls can be configured as speed dial or intercom calls. To configure a hotline button, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click on the **Administration > Users > Individual Users**.

Step 3 Click the name of the user whose phone you want to modify.

Step 4 Click the **Personal Options** tab.

Step 5 Click the **Program IP Phone Buttons** link. The Program IP Phone Buttons page appears as one shown in Figure 8-9 on page 214.

Step 6 In the IP Phone Type field, select the type of phone the user is using.

Step 7 Identify button # that you want to configure and do the following:

- **Step a** In the first Function field, select the **All** or **Telephony**.
- **Step b** In the second field, select Hotline.
- **Step c** In the Label field, enter a descriptive word to remind the user which function is associated with that button. This label can be up to five characters long for the multiline phones and up to six characters long for the IP 100 and BB24.The label will appear on the LED display adjacent to the button.
- **Step d** In the Extension field, enter the extension to which you want the call to connect.
- **Step e** In the Call Action field, select the method you want to use for making the connection.

Step 8 Click **Save**.

8.4.6.3 Copying Programmable Buttons Configurations

The Copy IP Phone Buttons page allows a system administrator to copy the programmable button configuration from one use to another, thus reducing the tedious work of configuring IP phone buttons.

To copy the programmable button configuration from one user's IP phone to another, follow the procedure below:

Step 1 Launch ShoreTel Director.

Step 2 Click on the **Administration > Users link > Individual Users**. The Individual Users page appears.

Step 3 Click on the name of the user whose IP phone you would like to modify. The Edit User page appears.

Step 4 Click the Personal Options tab.

Step 5 Locate the Program IP Phone Buttons link and click the **Copy** button. The Copy IP Phone Buttons dialogue box appears as shown in Figure 8-10.

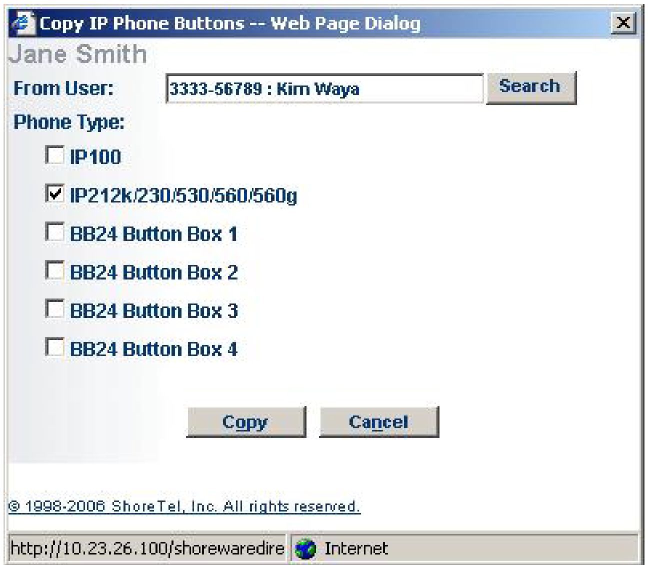


Figure 8-10 Copy IPPhone Buttons Dialog Box

Step 6 In the From User field, enter the name or extension of the party that you want the hotlink to call. You can enter the first letters or digits of the name or number of target party and click **Search** button to initiate search of the available users.

Step 7 In the Phone Type section, check appropriate phone option. The phone types must match, meaning that you cannot copy an IP110 button configuration to an IP230.

If both users have multiple phones on their desk (e.g. an IP560 phone with a BB24 button box), you can copy the button configuration for several devices simultaneously, assuming the other user also has an IP560 phone and a BB24 button box). To do this, just select the appropriate Phone Type check boxes. In this manner, you could copy up to four button box configurations at once.

Step 8 Click **Copy** to duplicate the programmable button configuration from one user to another.

To enable a user (as opposed to a system administrator) to configure the programmable buttons on his or her ShoreTel IP Phone, follow the procedure below:

Step 1 Launch ShoreTel Director.

Step 2 Click on the **Administration > Users link > Individual Users**.

Step 3 Click the name of the user whose profile you would like to modify (enabling him to customize the buttons on his IP phone). The Edit User page appears.

Step 4 Click the **Go to this User Group** link. The User Groups page for the user group to which the user is associated appears.

Step 5 For COS - Telephony, click **Go to this Class of Service**. The Class of Service page appears as shown in Figure 8-11.

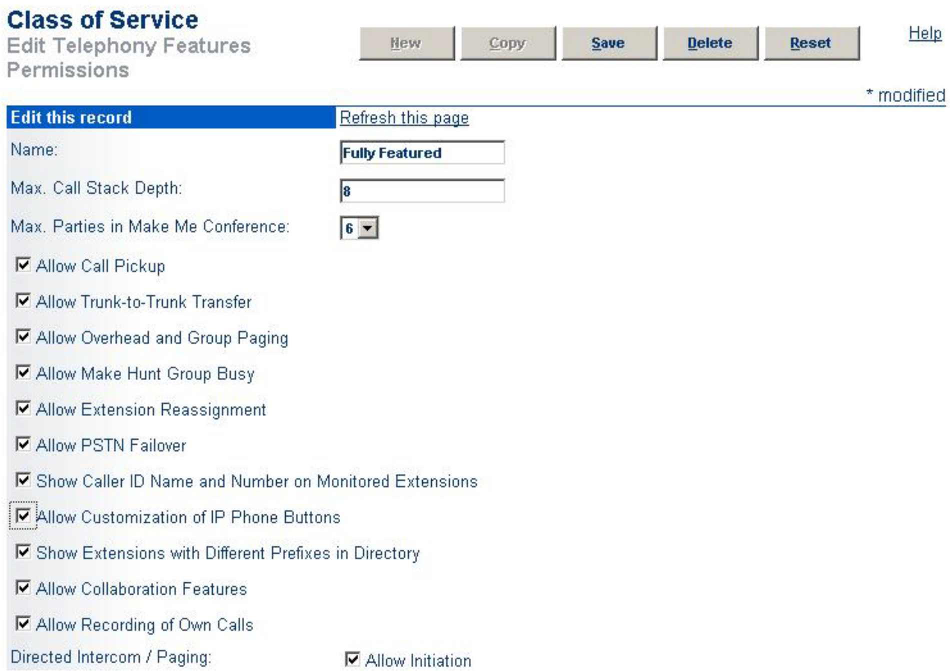


Figure 8-11 COS Edit Telephony Features Page

Step 6 Enter a name for the Class of Service profile in the Name field (unless you are modifying an existing Class of Service profile for which you would like to preserve the existing name).

Step 7 Check the **Allow Customization of IP Phone Buttons** check box.

Step 8 Click the **Save** button to store your changes.

Details:

- The default for new Class of Service (Telephony) profiles is to have this feature disabled, thus preventing users from modifying their own custom buttons.
- Once this Class of Service (Telephony) profile has been created, you can apply it to every person in the system that you would like to allow to change his or her custom buttons.

To change the custom buttons on your ShoreTel IP Phone or BB24 via the telephone interface, follow the procedure below:

Step 1 Press the **Options** button on your IP phone and enter your password, followed by the # key.

Step 2 Scroll through the list to option 4: Program Buttons.

Step 3 Press the **Edit** soft key.

Step 4 Press the custom button that you would like to modify. (If you are using the phone interface to modify the buttons on a BB24 device, press the button on the BB24 that you would like to configure.)

Step 5 Scroll through the list of functions until you find the function that you would like to apply to this button.

Step 6 When you have highlighted the appropriate function, press the ***Next*** soft key.

Step 7 Enter an extension, external number, or leave it blank. Then, press the ***Next*** soft key.

Step 8 Press the **1 > Aa** soft key to shift the key pad to alphabet mode.

Step 9 Use the key pad to enter a short descriptive word that will remind you of the new function of the custom button.

Step 10 Press the **Done** soft key.

Step 11 Press the **Done** soft key again to store your changes.

8.5 Malicious Call Trace

ShoreTel provides organizations with the ability to report a malicious call by requesting ShoreTel Communicator trace and record the source of the incoming call. Organizations deploying this release can provide their users with a method to initiate a sequence of events that will trace a call when they suspect that it has a malicious intent.

The Malicious Call Trace (MCT) feature enables the ShoreTel phone user to identify the source of malicious calls. A user, who receives a malicious call from the PSTN via an ISDN trunk supporting MCT, can initiate a MCT on the ShoreTel phone by pressing a programmable button or entering a star code sequence or using the Communicator toolbar button.

8.5.1 Implementation

Once the user initiates the Malicious Call Trace process, the ShoreTel Windows Event Log is notified and the user receives an urgent email confirming the action along with an audible tone. The system provider is notified through the PSTN of the malicious nature of the call. This allows the system provider to take action, such as notifying legal authorities.

NOTE Malicious Call Trace is an ISDN feature. It is implemented on BRI and PRI trunks to ISDN service providers supporting the feature. The ShoreTel implementation of MCT will support the ETSI standard configurable on switches supporting Euro-ISDN. Trace information is not provided to or displayed by the ShoreTel user phones.

8.5.1.1 Configuring a Programmable Button in Director

To configure malicious call trace button in ShoreTel Director, follow the procedure below:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Users > Individual Users**.

Step 3 Click on the name of the user whose phone you want to modify.

Step 4 Click the Personal Options tab.

Step 5 Click the **Program IP Phone Buttons** link. The Program IP Phone Buttons page appears as one shown in Figure 8-9 on page 214.

Step 6 In the IP Phone Type field, select the type of phone the user is using.

Step 7 Identify button # that you want to configure and do the following:

Step a In the first Function field, select the **All** or **Telephony**.

Step b In the second field, select **Malicious Call Trace**.

Step c In the Label field, enter a descriptive word to remind the user which function is associated with that button. This label can be up to five characters long for the multiline phones and up to six characters long for the IP 100 and BB24.The label will appear on the LED display adjacent to the button.

Step d In the Mailbox field, select the user mailbox to which you want to send the event logs.

Step e In the Call Action field, select the method you want to use for making the connection.

Step 8 Click **Save**.

8.5.1.2 Initiating a Malicious Call Trace with (*) Star Code

On a ShoreTel IP phone, SIP phone, analog phone, or Extension Assignment device the user needs to place the suspected malicious call on hold and then enter the MCT star code to start the tracing process.

Example: The user receives an incoming malicious call. Using a ShoreTel IP phone, SIP phone, Analog phone or Extension Assignment device the user presses the hold button and then enters *21 to start the trace sequence. Once the trace sequence starts, a confirmation tone will be played prior to returning to the call to indicate that an MCT request has been initiated, an event is logged in Director, record call is attempted to the local extension's mailbox, and an urgent email is sent to the recipient of the call.

8.5.1.3 Initiating a Malicious Call Trace using a ShoreTel IP Programmable Button

On a ShoreTel IP phone, the user presses the IP Programmable Button to start the tracing process.

Example: The user receives an incoming malicious call. Using a ShoreTel IP phone with programmable keys, the user presses the programmable key which will start the trace sequence. Once the trace sequence starts, a confirmation tone will be played to indicate that an MCT request has been initiated, an event is logged in Director, record call is attempted to the configured extension's mailbox, and an urgent email is sent to the recipient of the call.

8.5.1.4 Initiating a Malicious Call Trace using the Communicator Toolbar Button.

Using Communicator, the user presses the Communicator Toolbar button to start the tracing process.

Using Softphone, the user presses the IP Programmable Button to start the tracing process. The signal requesting MCT initiation is sent to the switch via TMS.

NOTE Communicator will not support the initiation of MCT using the star code.

Example: The user receives an incoming malicious call via Communicator or a softphone, the user can press the programmable toolbar key which will start the trace sequence. Once the trace sequence starts, a confirmation tone will be played to indicate that an MCT request has been initiated, an event is logged in Director, record call is attempted to the configured extension's mailbox, and an urgent email is sent to the recipient of the call.

Initiating a Malicious Call Trace using a ShoreTel IP Phone without Programmable Buttons or using Analog or SIP Phone

On a ShoreTel IP phone without programmable buttons or on Analog or SIP Phone, the user places the suspected incoming malicious call on hold then enters the MCT star code to start the tracing process.

Example: The user receives an incoming malicious call. The user places the call on hold and then enters *21 to start the trace sequence. Once the trace sequence starts, a confirmation tone will be played prior to returning to the call to indicate that an MCT request has been initiated, an event is logged in Director, record call is attempted to the local extension's mailbox, and an urgent email is sent to the recipient of the call.

8.5.2 Important Considerations

When setting up your system, keep the following in mind:

- ShoreTel MCT feature will only work with carriers supporting ETSI standard EN 300 130-1 V1.2.4.
- ShoreTel switches support the malicious call identification originating function (MCID-O) only. They do not support the malicious call identification terminating function (MCID-T). If the switch receives a notification from the network of a malicious call identification, it ignores the notification.
- Supported only for incoming calls from the ISDN network.
- Service provider must have MCID functionality enabled for the feature to work.
- ShoreTel ISDN interface on the ShoreTel E1/BRI switches must have the Protocol Type set to ISDN User with the Central Office Type set to Euro ISDN. When MCID is initiated on a SIP phone by putting a call on hold and initiating the star code *21 sequence, after the successful initiation of the signal the previous call continues to be held. User needs to manually unhold the call. For ShoreTel IP Phones and analog phones, the held call is connected back after MCT initiation.
- Malicious Call Trace confirmation tone is not given to SIP phone. SIP calls are not automatically taken off hold.
- Communicator and Softphone does not support initiation through the star code sequence.
- Mobile Communicator does not support the Malicious Call Trace feature.
- Malicious Call Trace confirmation tone signals an invocation attempt. It does not signal that the MCT request was successfully received at the connected network (CO). The MCT response is not processed by the ShoreTel ISDN stack.
- Malicious Call Trace phone programmable button may configure a target mailbox for recording the call, but MCT initiated via star code will always be recorded to the initiating user's mailbox (no way to specify target).
- Malicious Call Trace attempt can only be issued once per call.
- Malicious Call Trace invocation is only valid while the call is established.

- Malicious Call Trace is not supported on conference calls created on a ShoreTel PBX.
- Intercommunication/Networking considerations. MCID caller info on calls between different networks is subject to agreement between the service providers.

8.6 VPN Phone

VPN Phone provides secure audio communications through open VPN SSL tunnels for remote IP phones using the ShoreTel system. To support VPN Phone, ShoreTel offers two VPN Concentrators capable of supporting up to 100 calls through VPN tunnels. The ShoreTel IP Phones VPN Phone supports include the IP655, IP565g, IP560g, and IP230g.

Terms

Secure Socket Layer (SSL): A cryptographic protocol that provide secure communications on the Internet.

Point to Point Protocol (PPP): A data link protocol used to establish a direct connection between two nodes over serial cable, phone line, trunk line, cellular telephone, specialized radio links, or fiber optic links

Virtual Private Network (VPN): A computer network in which some internode links are facilitated via open connections or virtual circuits through a larger network instead of via physical wires. The link-layer protocols of the virtual network are said to be tunneled through the larger network. One common application is secure communications through the public Internet.

VPN Concentrator: A gateway that provides secure access to a corporate network for remote devices through VPN tunnels.

Stunnel: A multi-platform program that provide SSL tunnels between a remote device and a VPN gateway.

Tunnel: An link between two networks (such as LANs) that are connected by a larger network (such as a WAN or the Internet). Tunnel packets containing messages exchanged by the smaller networks are encapsulated into packets that facilitate routing through the larger network. Each VPN communication session establishes a tunnel, which is removed when the session is finished.

8.6.1 Functional Description

8.6.1.1 Feature Summary

VPN Phone provides secure audio communications for ShoreTel IP Phones located remotely from ShoreTel switches through open VPN SSL tunnels.

The feature includes an Open SSL VPN client in the ShoreTel IP phone and an Open SSL VPN Gateway. The Open SSL structure allows the traversal of firewalls implemented by many enterprises for blocking VPN tunnels.

8.6.1.2 Components

VPN Phone implementation requires two components – a VPN Concentrator and a ShoreTel IP Phone capable of communicating over a VPN

VPN Concentrator

The SSL based VPN Concentrator enables remote ShoreTel IP Phones to establish secure voice communications with through the local ShoreTel PBX through SSL VPN tunnels. For every tunnel, a virtual PPP interface is created on VPN Concentrator and a peer PPP interface is created on the remote ShoreTel IP Phone. Signaling and media streams go through the PPP interface and are secured by SSL encryption.

ShoreTel offers two VPN Concentrator Models:

- **VPN Concentrator 5300 –** supports a capacity of 100 calls.
- **VPN Concentrator 4500 –** supports a capacity of 10 calls.

The VPN Concentrator 4500/5300 Installation and Configuration Guide provides additional information about the ShoreTel VPN Concentrators.

VPN Phone Licenses

ShoreTel licenses VPN Phone usage on a stunnel basis. Establishing a stunnel uses a license, which becomes available when the tunnel is discontinued.

ShoreTel IP Phone

Three ShoreTel IP Phone models support the VPN Phone feature:

- IP565g
- IP560g
- IP230g

8.6.1.3 Network Configuration

Figure 8-12 depicts the functional structure the ShoreTel implementation of a VPN network.

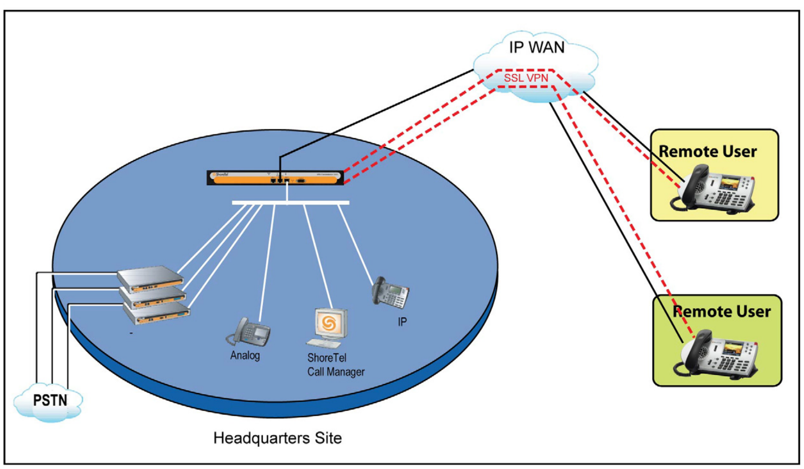


Figure 8-12 ShoreTel VPN Phone Implementation Architecture

The VPN Concentrator is located at ShoreTel's Main Site, connected to the same LAN as local ShoreTel switches and the Main ShoreTel server. Refer to the VPN Concentrator 4500/5300 Installation and Configuration Guide for specific deployment options based on the router and firewall configuration of the ShoreTel Network.

8.6.1.4 Stunnel Establishment

Each remote device is assigned a user name and password that is recognized by the VPN Concentrator. Phone logging into the Concentrator are authenticated through the verification of its user name and password. When the phone is successfully authenticated, the Concentrator establishes a stunnel to that phone, after which it can receive and make phone calls through the ShoreTel system. The stunnel remains in place until the phone sets the VPN parameter to off or the Concentrator times out all stunnel connections.

Establishing a stunnel requires an available VPN phone license. If the number of active stunnels equals the number of available licenses, The VPN Concentrator will not establish new stunnels until an existing stunnel is disconnected.

8.6.1.5 Phone Calls

After a stunnel is established from the ShoreTel IP Phone to the Concentrator, VPN phone calls are performed from the ShoreTel IP Phone in the same manner as if the phone is located on same LAN as the VPN Concentrator. The VPN Concentrator manages the connection from the phone to the ShoreTel.

8.6.2 Implementation

Configuring VPN Phone comprises three steps:

- Installing the VPN Concentrator
- Configuring Director
- Configuring the ShoreTel IP Phone

The following sections describe the VPN Phone implementation process.

8.6.2.1 VPN Concentrator

Refer to the VPN Concentrator 4500/5300 Installation and Configuration Guide for instructions on physically inserting the VPN Concentrator into the network. The guide also describes web browser pages that configure the VPN concentrator. The following sections describe the pages and fields that require configuration.

Initial Configuration

The VPN Concentrator is shipped with the pre-configured IP address 192.168.1.1 for the LAN port.

To access the VPN Concentrator web interface pages, perform the following:

Step 1 Assign static IP address 192.168.1.2 with subnet 255.255.255.0 to the Ethernet interface of the computer that is connected to the LAN port.

Step 2 Launch a web browser on the PC and access the following URL:

- `http://192.168.1.1.`

The computer displays the login page shown in Figure 8-13.

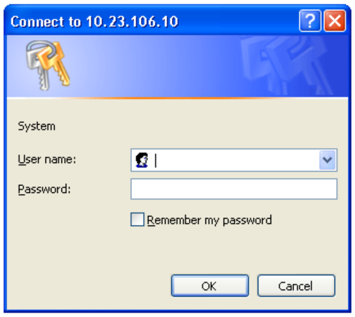


Figure 8-13 VPN Concentrator Login Page

Step 3 Enter the following parameter values to log into the system:

- **Username** – root
- **Password** – default

Network Page

The Network page defines the IP address through which the VPN Concentrator is accessed. To access the Network page, select Network in the blue Configuration Menu on the left side of the page. Figure 8-14 displays the Network page.

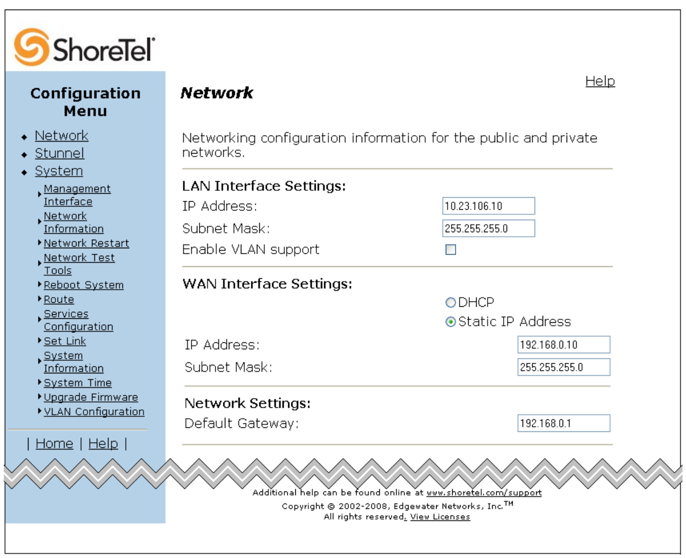


Figure 8-14 Network Configuration Page

Enter the appropriate values in the following data entry fields:

- LAN Interface Settings: Enter the IP address by which other LAN devices will access the VPN Concentrator
- WAN Interface Settings: Enter the IP address by which remote devices can access the VPN Concentrator.

Stunnel Page

The Stunnel page specifies setup characteristics about the Stunnels that facilitate communications between the concentrator and the remote phones. To access the Stunnel page, select Stunnel in the blue Configuration Menu on the left side of the page. Figure 8-15 displays the Network page.

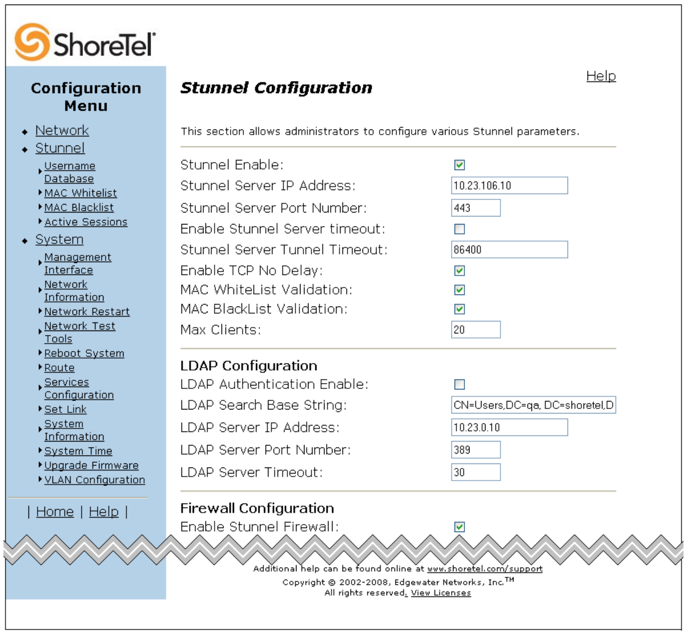


Figure 8-15 Stunnel Configuration Page

Verify that the following parameters are set as follows:

- Stunnel Enable is selected.
- Stunnel Server IP Address is set to the LAN address of the VPN Concentrator, as specified on the Network page.
- Stunnel Server Port Number is set to 443.

Route Page

The Route page defines the static routes to networks or servers on the LAN. To access the Route page as shown in Figure 8-16, select System > Route in the blue Configuration Menu on the left side of the page.

Figure 8-16 Route Configuration Page

To add a static route, do the following:

Step 1 Enter the subnet address and mask in the IP Network and Netmask data fields, respectively.

Step 2 Enter the IP address that accesses the Gateway server of the added network.

Step 3 Press the Submit button.

Usernames' List Page

The Username Configuration page defines the user accounts that are allowed Stunnel access to the VPN Concentrator. To access the Username's List page as shown in Figure 8-17, select Stunnel > Username Database in the blue Configuration Menu on the left side of the page.

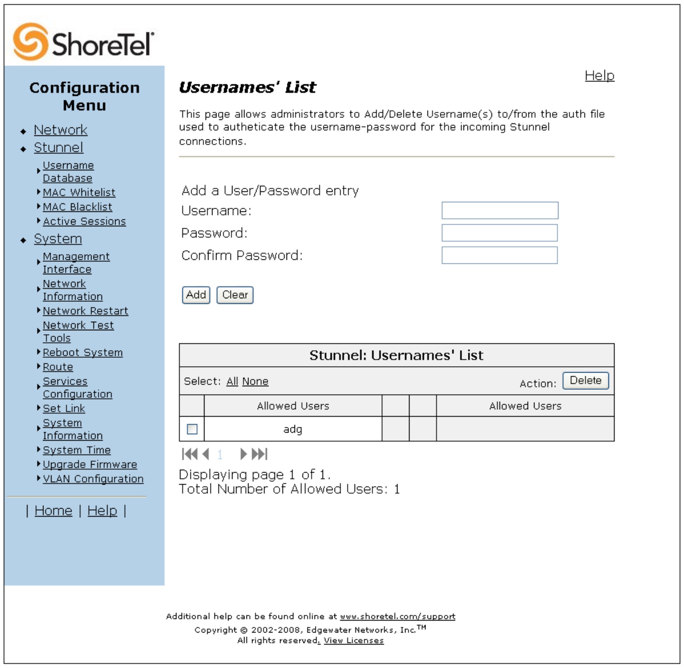


Figure 8-17 Username Configuration Page

To add a user account, perform the following:

Step 1 Enter the username and password for the user in the Username and Password data fields, respectively.

Step 2 Re-enter the password in the Confirm Password data entry field.

The Password and Confirm Password data entry field contents must be identical.

Step 3 Press the **Submit** button.

8.6.2.2 ShoreTel Director

ShoreTel Director assigns codecs on the basis of the site assignment of the ShoreTel IP Phone's IP address. Assigning the IP address block allocated to the VPN Concentrator to a specific site assures that the switch uses the proper codec when handling VPN Calls.

To assign an IP address block to a site, perform the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > IP Phones > IP Phone Address Map**. The IP Phone Address Map List page appears.

Step 3 Click the **New** button. The IP Phone Address Map Info appears as shown in Figure 8-18.

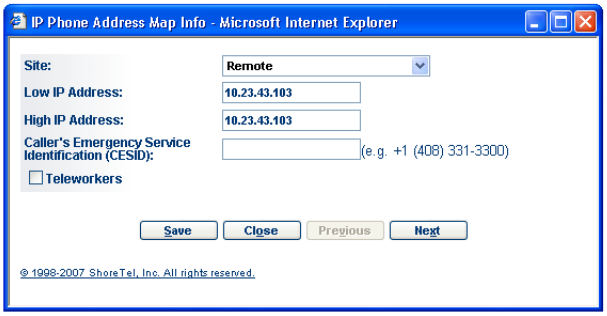


Figure 8-18 IP Phone Address Map Info Page

Step 4 In the Site Name field, select the VPN site.

Step 5 In the Low IP Address field, enter the lowest IP address of the block allocated to VPN calls. The IP address must be valid for the network where the site is located.

Step 6 In the Low IP Address field, enter the highest IP address of the block allocated to VPN calls.

Step 7 Press the **Save** button.

NOTE Refer to Table A-1 on page 603 for configuration recommendations for Emergency 911 related features on VPN phones.

8.6.2.3 Phones

ShoreTel IP Phones are manually configured to establish a stunnel with the VPN concentrator. After the phone is configured and placed on a WAN port (such as the internet) it attempts to communicate with the Concentrator. ShoreTel does not provide DHCP options for automatically setting these parameter values at startup.

The following parameters are manually configured on the ShoreTel IP Phone to perform VPN calls:

- VPN Gateway: This parameter specifies the WAN IP Address of the VPN Concentrator to which the ShoreTel IP Phone connects. Default value is 0.0.0.0.
- VPN Port: This parameter specifies the port number of the VPN Concentrator to which the ShoreTel IP Phone connects. Default value is 443.
- VPN: This parameter, when set to On, enables VPN Phone on the ShoreTel IP Phone. Default setting is Off.
- VPN User Prompt: This parameter, when set to On, programs the ShoreTel IP Phone to prompt the user for a VPN user name after completing a power cycle.
- VPN Password Prompt: This parameter, when set to On, programs the ShoreTel IP Phone to prompt the user for a VPN password after completing a power cycle.
- FTP: This parameter specifies the IP Address of the RTP server from which the phone requests VPN Phone software upgrades. When set to the default value of 0.0.0.0, the phone solicits upgrades from the IP Address of the VPN Gateway.

Manually Configuring a ShoreTel IP Phone

To configure the ShoreTel IP Phone IP phone, perform the following:

Step 1 Press the Mute button, then enter SETUP#

SETUP translates numerically to 73887

Step 2 Enter the phone's password, followed by #.

Step 3 Press the # button to step through the phone options.

Configuring the User Name and Password

The user name and password is stored in non-volatile RAM on the phone. Power cycling and normal phone operations have no effect on the stored name and password. The VPN Concentrator authenticates the ShoreTel IP Phone when the phone attempts to establish a stunnel by verifying the phone's username and password is included in the user accounts on the Usernames' List page.

New ShoreTel IP Phones are shipped with this memory location vacant. The first time a user power cycles the phone with the VPN parameter set to On, the phone prompts the user for a username and password. The phone prompts for these values if the VPN User Prompt and VPN Password Prompt parameters are set to On; otherwise, the ShoreTel IP Phone continues using the previous memory contents when attempting to establish a stunnel.

8.7 Simultaneous Ringing and Call Move

Simultaneous ringing allows a ShoreTel user to ring up to 2 additional phones in addition to their assigned phone. The simultaneous ringing configuration is done through the user's Communicator interface or from their user page in ShoreTel Director.

When the feature is configured, calls to the ShoreTel extension of the user ring the primary phone and all additional configured phones simultaneously. For convenience, the feature can be turned on and off to stop the simultaneous ringing at any time.

Incoming calls to simultaneous ringing devices are presented as standard calls with standard ring tone. A ring delay can be configured for additional destinations which allows the preferred phone to ring first.

After a simultaneous ringing call is established, the call may be moved between the simultaneous ringing devices. The Call Move mechanism can be initiated by a ShoreTel IP Phone softkey button, the Communicator call cell button, or a star code sequence (*23).

8.7.1 Implementing Simultaneous Ringing and Call Move

Administrators can enable the features through the Telephony Class of Service. Any ShoreTel User may be configured for Simultaneous Ringing. After a ShoreTel User is configured for Simultaneous Ringing, their extension becomes Preferred. This preferred user can be configured as a standard system extension, external Extension Assignment, SoftPhone, VPN phone, SIP phone, or Analog phone.

The process for configuring Simultaneous Ringing and Call Move is described in the next sections.

8.7.1.1 Modifying Class of Service

In order for the user to be able to configure Simultaneous Ringing of their phones, the Administrator must first modify the default Class of Service in order to enable the user. To modify the Class of Service, follow these steps:

Step 1 Log onto ShoreTel Director.

Step 2 Click **Administration > Users > Class of Service**.

Step 3 In the Telephony Features Permissions section, select an existing feature to modify or click Add New to create a new Class of Service. The Edit Telephony Features Permissions page appears.

Step 4 Check the **Allow External Call Forwarding and Find Me Destinations** check box as shown in Figure 8-19. and option.

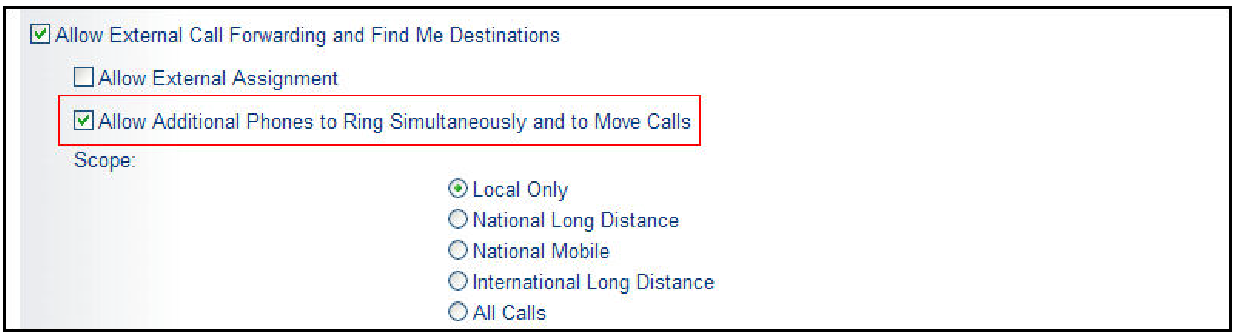


Figure 8-19 Class of Service - Edit Telephony Feature Permissions

Step 5 Check the **Allow Additional Phones to Ring and Move Calls** check box.

Step 6 In the Scope section, click the radio button to specify the type of calls on which you want to allow users of this class of service to use these features.

Step 7 Click **Save** to save your configuration.

8.7.1.2 Configuring Users

To configure users for simultaneous ringing, follow these steps:

Step 1 Log into ShoreTel Director.

Step 2 Click the **Administration > Users > Individual Users**. The Individual Users page appears.

Step 3 Select the user profile that you want to modify or create a new user.

Step 4 Click the **Personal Options** tab.

Step 5 Click **External Assignment and Additional Phones**. The Find Me, External Assignment and Additional Phones page as shown in Figure 8-20 appears.

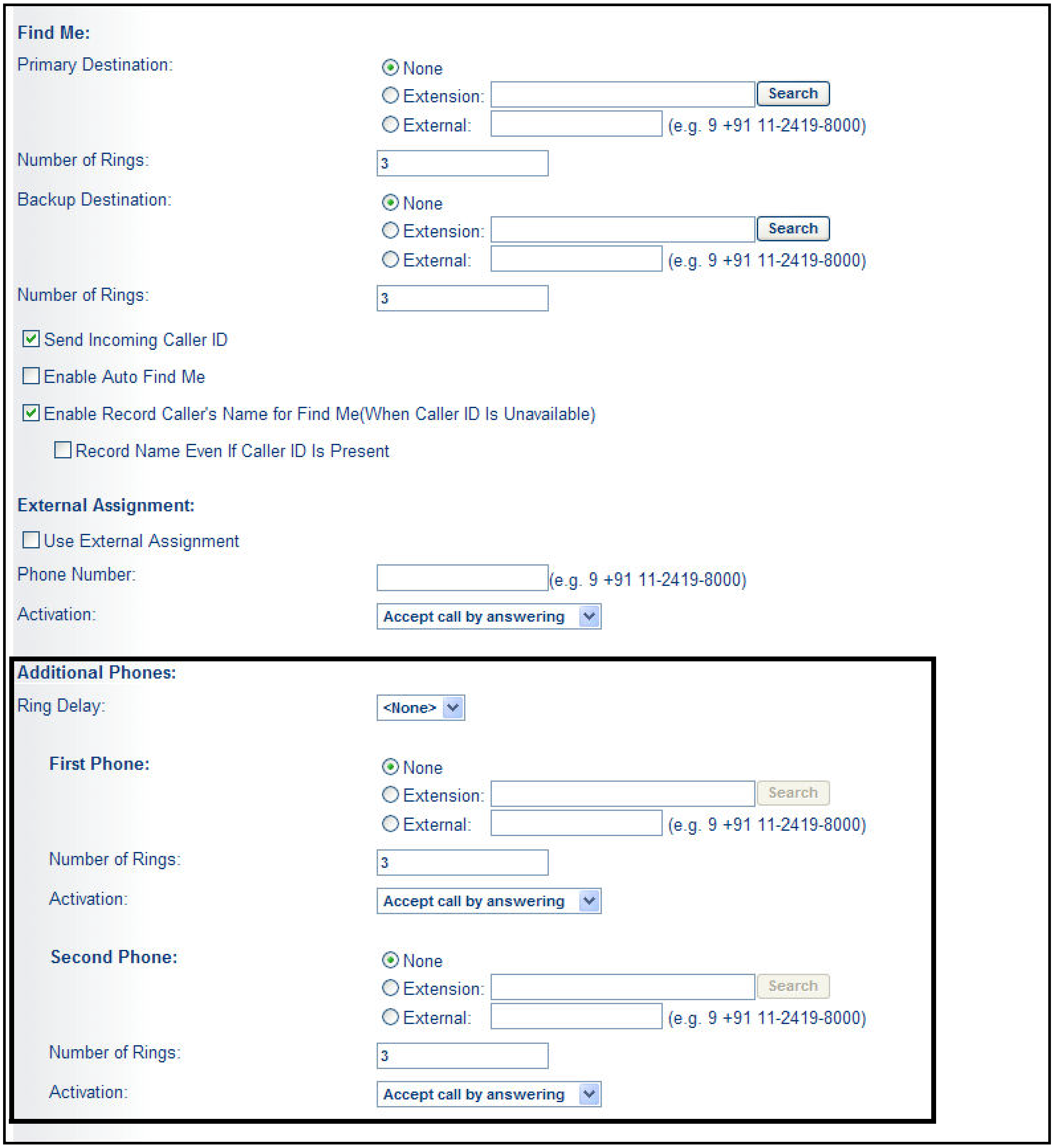


Figure 8-20 Find Me, External Assignment and Additional Phones Page

Step 6 In the Additional Phones section, do the following to configure a first and second phone as required:

Step a In the Phone section, click a radio button to indicate the additional extension you want to ring as follows:

- **None**: No number.
- **Extension:** Enter or select a ShoreTel extension in the field.
- **External:** Enter the dialable number of an external phone in the field.

Step b In the Number of Rings field, enter the number of times you want the phone to ring before the call is rerouted.

Step c In the Activation field, select the method the user is to use to answer calls:

- **Accept call by answering:** Requires the user to remove the phone from the hook and speak.
- **Accept call by pressing '1':** Requires the user to press 1 on the phone to signal that they are answering.

Step 7 Click Save to save your configuration.

8.7.1.3 Configuring Simultaneous Ringing through Communicator

Simultaneous Ringing and Call Move can be configured through the ShoreTel Communicator Options and Preference page. Users can access this page to configure the additional phones they want to use for simultaneous ringing. This will allow users to quickly turn Simultaneous Ringing On or Off as needed.

To configure Communicator to use this feature, follow these steps.

Step 1 Launch ShoreTel Communicator.

Step 2 Click **Tools > Options**. The Options and Preferences page appears.

Step 3 Click Addition Phones. The Additional Phones page appears as shown in Figure 8-21.

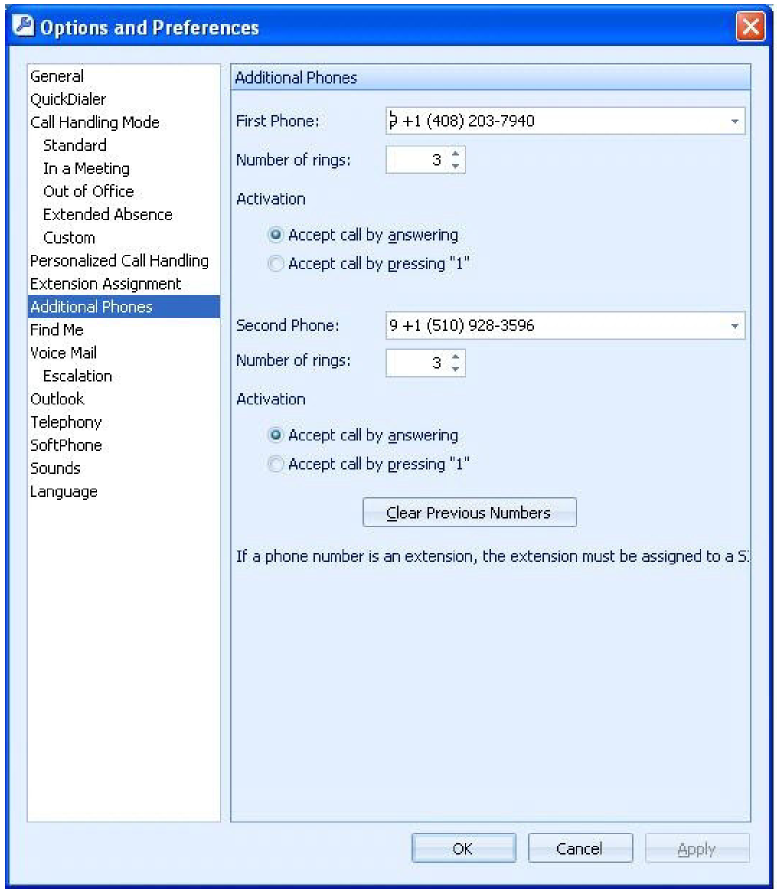


Figure 8-21 Options and Preference - Additional Phones Page

Step 4 Do the following to configure a first and second phone as required:

Step a In the Phone section, click a radio button to indicate the additional extension you want to ring as follows:

- **None:** No number.
- **Extension:** Enter or select a ShoreTel extension in the field.
- **External:** Enter the dialable number of an external phone in the field.

Step b In the Number of Rings field, enter the number of times you want the phone to ring before the call is rerouted.

Step c In the Activation section, click the radio button to specify the method you want to use to answer calls:

- **Accept call by answering:** Requires the user to remove the phone from the hook and speak.
- **Accept call by pressing '1':** Requires the user to press 1 on the phone to signal that they are answering.

Step 5 Click **Save** to save your configuration.

8.7.1.4 Disabling/Enabling Simultaneous Ringing

Users can enable or disable the simultaneous ringing functionality through their Communicator application and allow them to quickly turn simultaneous ringing on or off as needed.

To enable or disable simultaneous ringing, follow these steps:

Step 1 Launch ShoreTel Communicator.

Step 2 Click the ShoreTel logo. A drop-down menu appears.

Step 3 Highlight **Additional Phones**. A drop-down menu appears (see Figure 8-22).

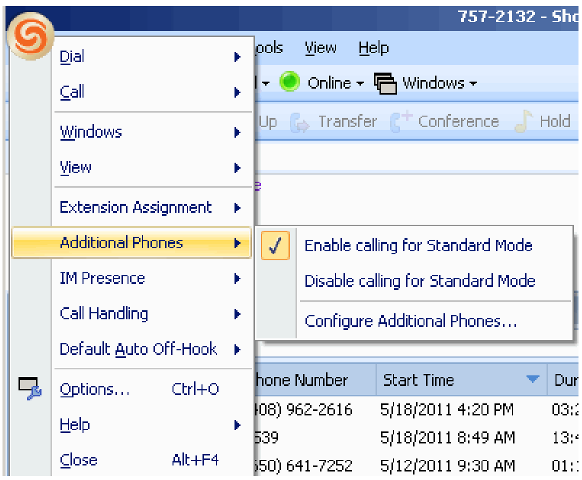


Figure 8-22 Communicator - Additional Phones Page

Step 4 Do one of the following:

- **Select Enable calling for...**: Turn on this feature for the current call handling mode.
- **Disable calling for...**: Turn off this feature for the current call handling mode.

The status of the ShoreTel extension changes immediately.

8.7.2 Using Call Move

The ShoreTel system supports call move. Call move allows users to switch from one phone device to another without disrupting the conversation. For instance, if you attend a conference call, you easily move your call from your desk phone to your cell phone and go, without disrupting the conference.

Call Move from an IP Phone or Communicator

The user sees two behaviors while moving a call:

- When the call is on the assigned phone and the user presses MOVE or uses the Move Call Action from Communicator, the call goes on hold. Simultaneous ringing on idle phone(s) lasts until the user picks up one the phone. Until the user picks up the call, the caller hears silence.

- When the call is on one of the Additional Phones and you press MOVE on the assigned phone or use the Move Call action from Communicator, the conversation is immediately switched to the assigned phone
  - — The preferred phone pushes the softkey Move or Move Call action from Communicator button to move the call to an Additional Phone. NOTE: Preferred phone will not ring.
  - — Additional destination phones will start ringing with no ring delay.
  - — Ringing on Additional phones will stop after the call is answered by an Additional phone.
  - — When additional answers, Call is Moved to Additional destination.

8.7.2.1 Call Move Using Star Code on a Mobile Phone

- Dial *23 (at the dial tone).
- Your simultaneous ringing- idle- phone(s) will ring until you pick one of them.
- Until then, the caller will hear silence.
- Additional destinations will ring.
- When the call is answered on the additional phone, the call is moved to new device.

8.7.2.2 Call Move Cancellation

- Unhold the call or execute *23 code again.
- If the Cancel is successful, the call is retrieved.

8.7.2.3 Other Considerations

- Only SIP extension type phones can be configured as an additional device.
- If a conference call is in progress Call Move operation is not allowed.
- Call Move pull functionality is not supported from additional destinations.
- Call Move push functionality is not supported on SIP trunks only if they support DTMF signaling using SIP INFO.
- Call Move pull functionality is not supported from Communicator.
- The only supported star code sequences from additional destinations is *23.
- Communicator only displays calls on assigned phone.
- Call Move or Simultaneous Ringing on/off is not supported from ShoreTel Mobile Communicator.
- When OSE is configured as additional phone, care should be taken to make sure the call is directly placed to OSE and not AA.
- When cell phone is configured as additional phone, care should be taken to set the activation mode to 'answer by pressing 1' so that when the call is redirected to cell phone VM, other simultaneous ringing destinations do not stop ringing.
- If the preferred user is a work group agent, WrapUp softkey will be presented in place of AddOn/ AddOff softkey. However if the user receives a personal call (not a WorkGroup/ Contact Center call) then MOVE softkey will be presented.

CHAPTER 9

Setting Call Control Options

This chapter provides information about configuring the system-wide call control features of the ShoreTel system. The topics include:

9.1 Account Codes

Account codes are typically used to assist ShoreTel users in the billing of their clients. For example, if a law firm wants to keep track of the length of calls to their clients so that they may later bill those client for services rendered, they can enter an account code that corresponds to that client before dialing the client's phone number. At the end of the call, the call length, time, and date are entered in a record, thus helping the firm to keep track of the calls made to each of their clients.

Account Codes can vary in length and be flexibly formatted. You can configure the system to require users to enter an account code for outbound calls in a mandatory or optional fashion. In this way, the account code can also function to prevent unauthorized employees from dialing long-distance numbers.

ShoreTel supports wildcard characters in account codes. This enhancement allows the system to surpass the previous limit of 50,000 account codes so that an almost unlimited number of account codes can be supported. The wildcard character – a question mark – can be entered in place of DTMF digits in the account code.

The use of wildcards introduces less strict validation of the account code entered by the user. Rather than checking each individual code, a length check is performed instead. The introduction of wildcards into the account codes does not impact the ability of the system to assign an account code to individual clients.

You can create account codes with non-numeric characters, but these characters are discarded during code collection. The following table gives example account codes and shows how the Account Codes Service interprets the code.

Table 9-1 Sample Account Codes

Sample Account Code	Recorded Code
Sales 200	200
1001-3	10013
1.234A	1234
3000 Exec 2	3002

Account code collection is enabled on the basis of user groups, with the collection of account codes set to be one of three states:

- Disabled
- Optional
- Forced

For information about account codes and User Groups, refer to Section 10.3.

Call Detail Reports include details of account codes associated with outbound calling. Account Codes are associated with a configurable extension and have a dedicated user group that defines ultimate call permissions and trunk group access.

A new user group named “Account Codes Service” is created for use by the Account Codes Service. Since it is only intended for use by the Account Codes Service, this user group does not appear in drop-down lists for the assignment of User Groups to users and other objects such as workgroups. You can, however, change all attributes of the Account Codes Service User Group except the fields indicating whether Account Codes are disabled, optional, or required.

9.1.1 Account Code Collection

When account code collection is enabled or required for a user group, calls placed via the telephone or the Communicator are routed to the account code extension. The Account Codes Service prompts the user to enter an account code followed by the “#” key. If the account code entered does not match the digits in a stored account code, an explanation message is played and the user can enter an account code again. When a matching account code is collected, the call is placed according to the originally dialed number.

The call permissions define which dialed numbers will be directed to the Account Codes Service for user groups configured with account codes. For calls that are redirected to the account codes extension, the call will be completed with the trunk access and call permissions of the Account Codes Service.

This structure imposes two sets of permissions to outbound calls:

1. The call permissions for the user group of the user that places the call are used to determine whether an account code must be collected or not.
2. The call permissions for the Account Codes Service user group determines whether calls are finally placed or if the intercept tone is played.

Calls forwarded to external numbers are NOT subjected to account code restrictions. Forwarding calls to external numbers is controlled from the Class of Service settings. For more informations, see the "Classes of Service" on page 314.

The Account Codes Service is associated with a system extension that is hosted on the SoftSwitch running on the headquarters (HQ) server only. If the HQ SoftSwitch is not reachable by the originating ShoreTel switch, the call will be handled according to the setting on the caller's user group. Specifically, during such a connectivity outage, calls placed by users who have optional account code collection will be automatically placed and calls placed by users who have forced account code collection will be automatically rejected.

If you have a ShoreTel Converged Conference solution installed, Account Codes can be set up for collaborative voice and data conferencing. Document sharing is a feature allowed in a 2-way or Make Me Conference call. Up to 20 documents can be dragged and dropped during the call. You can create associated project codes on the conference bridge for tracking purposes. Bridge project codes are not enforced.

9.1.2 Adding and Editing Account Codes

To configure the account code wildcard feature follow the procedure below:

Step 1 Launch **ShoreTel Director**.

Step 2 Click **Administration > Call Control > Account Codes**. The Account Codes page appears as shown in Figure 9-1.

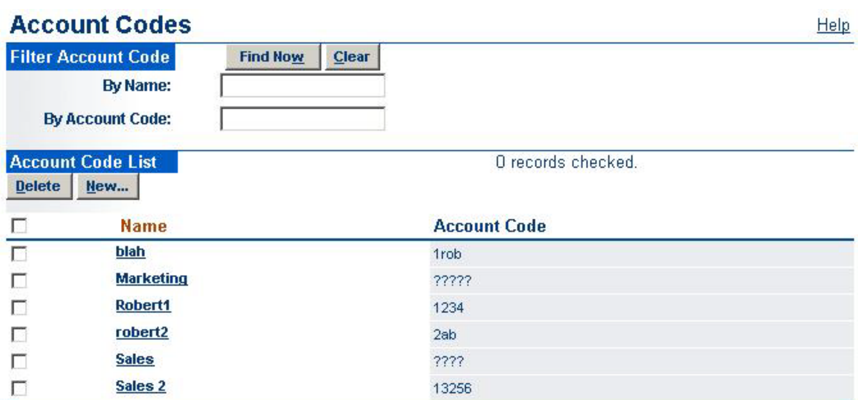


Figure 9-1 Account Codes Page

NOTE The Filter Account Code section lets you search for existing account code by name or account code. To search, enter the beginning string of the name or account code in the Name or Account Code field and click Find Now. To display the entire list, leave both fields blank and click Find Now.

Step 3 Click the **New** button. The Account Code Info dialog box opens (Figure 9-2).

Figure 9-2 Account Code Info Dialog Box

Step 4 In the **Name** field, enter a name for the account code.

Step 5 In the Account Code field, enter an account code.

Account codes can include up to 20 alpha-numeric characters and must include at least one digit. Digits are significant characters and may not be replicated as a particular string. For example, the system identifies 8888, p88q88, and abc8de8fg8hij8 as the same code: 8888.

Step 6 Click **Save** to create the account. To create another account, click ***Next***.

Account codes are enabled on the basis of user groups. For more information, see the "User Groups" on page 325.

9.2 Multi-site Account Codes

Many organizations need to track calls made to clients so that they can bill the clients for the time they spend on the call. The Multi-site Account Codes feature allows the ShoreTel Headquarters server, a Distributed Voicemail Server, or a ShoreTel Voicemail Switch to validate account codes. This feature benefits customers by distributing the processing of the account code validation load thus eliminating any single point of failure for account codes validation. Account Codes is an easy to use tool that helps businesses identify and invoice for client calls.

Account code validation is performed by the Headquarters server by default. Outbound external calls are redirected to the Headquarters server for account code validation. Once the account code is validated, the call is redirected to the originally dialed external number.

The Multi-site Account Codes feature adds the capability of allowing a Distributed Voicemail Server or ShoreTel Voicemail Switch to validate account codes. In addition, if the Distributed Voice Server or ShoreTel Voicemail Switch is unavailable, the Account Code validation will migrate to another ShoreTel Server or ShoreTel Voicemail Switch according the site hierarchy. This allows account code validation to be more reliable in a multi-site environment.

9.2.1 Implementation

9.2.1.1 Configuring Multi-site Account Codes

To successfully configure your ShoreTel system with Multi-site Account Codes, you must complete the following configuration activity:

- Change the System-wide Account Code Extension
- Change the Account Codes Local Extension for the Headquarters Server
- Add the Account Codes Local Extension to the Distributed Voicemail Server
- Add the Account Code Local Extension to the ShoreTel Voicemail Switch

9.2.2 Usage

To use Account Codes, a person picks up the phone and dials an external number. The call is redirected to a ShoreTel Server or ShoreTel Voicemail Switch and they are prompted to enter an account code. The person enters the account code and presses the # key. If the user enters a valid account code, the call is recorded in the database and redirected to the external destination. Then ShoreTel system administrator can then run Account Code reports to show the time, date and length of the call.

If the person repeatedly dials an incorrect account code, and the ShoreTel system administrator has configured the callers User Group such that it is mandatory to enter an account code (Forced), the call will be dropped.

If the person repeatedly dials an incorrect account code, and the ShoreTel system administrator has configured the callers User Group such that it is optional to enter an account code (Optional), the call will be proceed to the external number, but it will not be recorded in the database and made available to Account Code reports.

9.2.3 Configuring Multi-site Account Codes

9.2.3.1 Changing the System-wide Account Code Extension

The System-wide Account Code Extension is populated by default when the ShoreTel software is installed on the Headquarters server. If the system-wide Account Code Extension conflicts with the existing customer dial plan, the administrator can change the Account Code Extension.

To change the System-wide Account Code Extension, follow these steps:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > System Parameter > Systems Extensions**. The System Extensions edit page appears as shown in Figure 9-3.

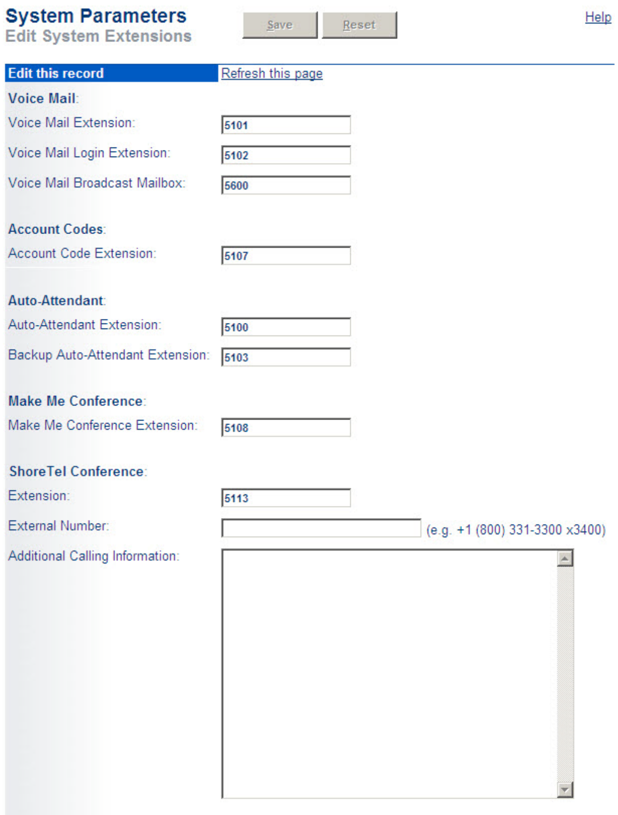


Figure 9-3 System-wide Account Code Extension Page

Step 3 In the Account Code Extension field, change the extension.

Step 4 Click Save button to save your changes.

9.2.3.2 Changing the Account Code Local Extension for the Headquarters Server

The Account Code Local Extension is populated by default for the Headquarters Server only.

To change the Account Code Local Extension for the Headquarters server, follow these steps:

Step 1 Log into ShoreTel Director.

Step 2 Click on the **Administration > Applications Servers > HQ/DVS**. The HQ/DVS Servers page appears.

Step 3 Select your headquarters server. The HQ/DVS Edit Servers page for the headquarters server appears as shown in Figure 9-4.

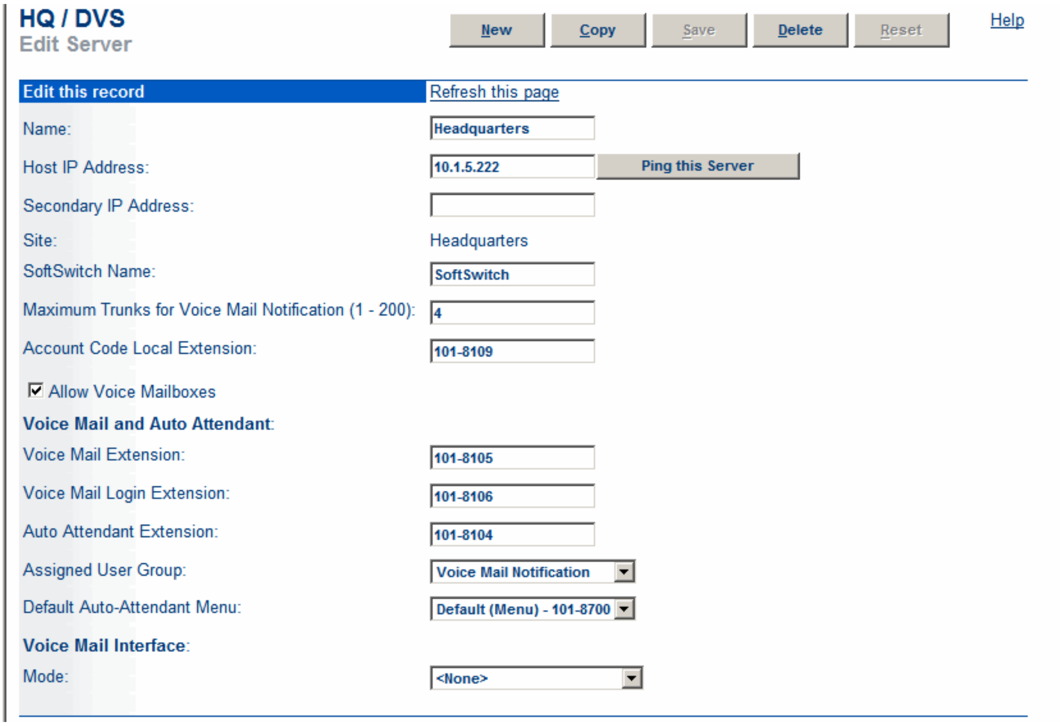


Figure 9-4 HQ/DVS Edit Server Page for Headquarters Server

Step 4 In the Account Code Local Extension field, enter the account code that you want to use for the headquarters server.

9.2.3.3 Adding the Account Code Local Extension to the Distributed Voicemail Server

The Account Code Local Extension is not populated by default for Distributed Voicemail Servers (DVS). The administrator must manually enter the extension.

To add the Account Code Local Extension to a Distributed Voicemail Server, follow these steps:

Step 1 Log into ShoreTel Director.

Step 2 Click on the **Administration > Applications Servers > HQ/DVS**. The HQ/DVS Servers page appears.

Step 3 Select the Distributed Voicemail Server to which you want to assign an account code. The HQ/DVS Edit Servers page for the DVS server appears similar to that shown in Figure 9-4 above.

Step 4 In the Account Code Local Extension field, enter the account code that you want to use for the DVS server.

Step 5 Click on the **Save** button to save your changes.

9.2.3.4 Adding the Account Code Local Extension to the ShoreTel Voicemail Switch

The Account Code Local Extension is not populated by default for Voicemail Switches. The administrator must manually enter the extension. To add the Account Code Local Extension for a ShoreTel Voicemail Switch, follow these steps:

Step 1 Log into ShoreTel Director.

Step 2 Click **Platform Hardware > Voice Switches/Service Appliances > Primary**. The Primary Voice Switches/Service Appliances page appears.

Step 3 Select a ShoreTel voicemail switch (50V or 90V) to which to add an account code local extension. The Edit Voice Switches page for the voice switch appears as shown in Figure 9-5.

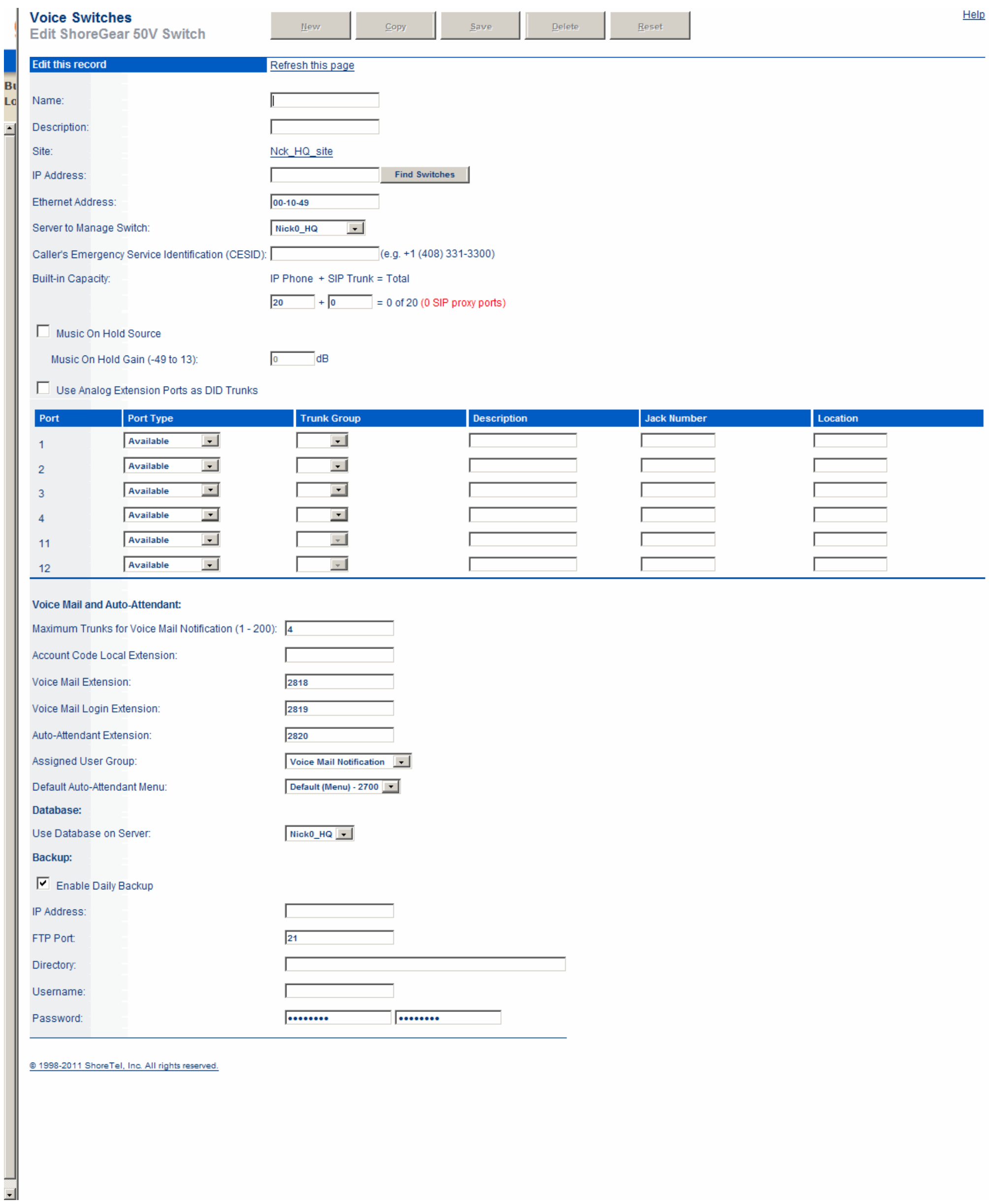


Figure 9-5 Edit Voice Switch Page for Voicemail Switch

Step 4 In the Account Code Local Extension field, enter an account code for local extensions.

Step 5 Click **Save**.

Additional Configuration Requirements

Once the system has been configured to validate account codes, the ShoreTel system administrator must restrict the users call permissions so that outbound external calls are not permitted.

9.3 Bridged Call Appearances

A Bridged Call Appearance (BCA) is an extension that is shared among multiple users. A BCA has an internal extension number and call stack depth. Each user with the same BCA sees the same BCA extension number, and the number of calls that can reside on that BCA is its call stack depth. The maximum BCA stack size is 24.

Some characteristics of a BCA are as follows:

- It does not require a license.
- It is assigned to a ShoreTel Voice Switch.
- It supports Call Handling Mode.
- It cannot be controlled by a schedule.

A user answers a BCA call by pressing a ShoreTel IP Phone button that is assigned to a BCA call stack position. Calls to a BCA occupy distinct call appearances and are identified by their position in the call stack. ShoreTel IP Phone buttons that answer calls are configured to handle calls to a specific call stack position of a BCA. A button can be programmed for each position in the call stack.

Example: The system administrator configures a BCA with an extension of 118 and a call stack depth of 3. The ShoreTel IP Phones of three users are configured to handle calls to the BCA as follows:

- User One has one button that answers calls from Stack Position #1.
- User Two has one button that answers calls from Stack Position #2.
- User Three has three buttons configured to answer BCA calls – the first button answers calls to Stack Position #1, the second button answers calls to Stack Position #2, and the third button answers calls to Stack Position #3.

The first incoming call to the BCA arrives on Stack Position #1. User Two cannot answer this call. A second call to the BCA will arrive on Stack Position #2 if the first call is still active. User One cannot answer that call. User Three is the only user that can answer calls that arrive on Stack Position #3.

When a call stack position on a BCA receives a call, the button on each phone configured for that stack position flashes orange to indicate an incoming call. When the call is answered, the LED on the phone of the person that answers it turns solid green while the other BCA stack buttons are red (without BCA conferencing) or orange (when BCA Conferencing has been enabled for the BCA).

A user places a call from a BCA by pressing a programmed ShoreTel IP Phone button. The LED on the outbound caller's phone becomes solid green, and the buttons associated with the BCA stack position on all other phones become solid red. If the call is placed on hold, the button LED for the applicable call stack position on all phones indicates a call on hold.

Pressing the top-most BCA custom button for outbound calls does not necessarily access trunk 1. No one-to-one correlation exists between the custom buttons programmed for BCA extensions and a particular trunk. The system administrator can associate trunks with BCA extensions through a variety of approaches.

A caller ID number can be associated with a BCA. The following rules determine which caller ID number is displayed at the far end for an outbound BCA call:

- Outbound to an internal extension – the name and number of the user that initiated the BCA call is sent. If the user is a "private," the caller ID is blank.
- Outbound to an external number – the system sends the first number in the following list that is available:
  - — Outbound caller ID number that is assigned to the BCA
  - — DID number assigned to the BCA
  - — External identification or caller ID number of the user who initiates the BCA call
- Outbound to an external emergency number (such as 911) – the emergency identification or the user's CESID number is sent.

The system can be configured to display the caller ID on inbound calls. It also can be configured to enable, disable, or delay inbound call ringing.

9.3.1 Switch Support for Bridged Call Appearances

ShoreGear one-rack unit (1-U) Half Width and 1-U Full Width voice switches support BCAs with the following limits:

- Up to 24 BCA extensions can be configured on a switch.
- Up to 128 BCA extensions (on other switches) can be monitored.
- A maximum of 32 phones can be configured to point to the same BCA extension.

9.3.2 Configuring BCA Parameters

To create a BCA profile:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Call Control > Bridged Call Appearances** from the Main Menu. The Bridged Call Appearances List page appears as shown in Figure 9-6.

Figure 9-6 Bridge Call Appearance List Page

Step 3 Click **New** (or select an existing BCA profile to edit). The Edit Bridged Call Appearances page appears as shown in Figure 9-7.

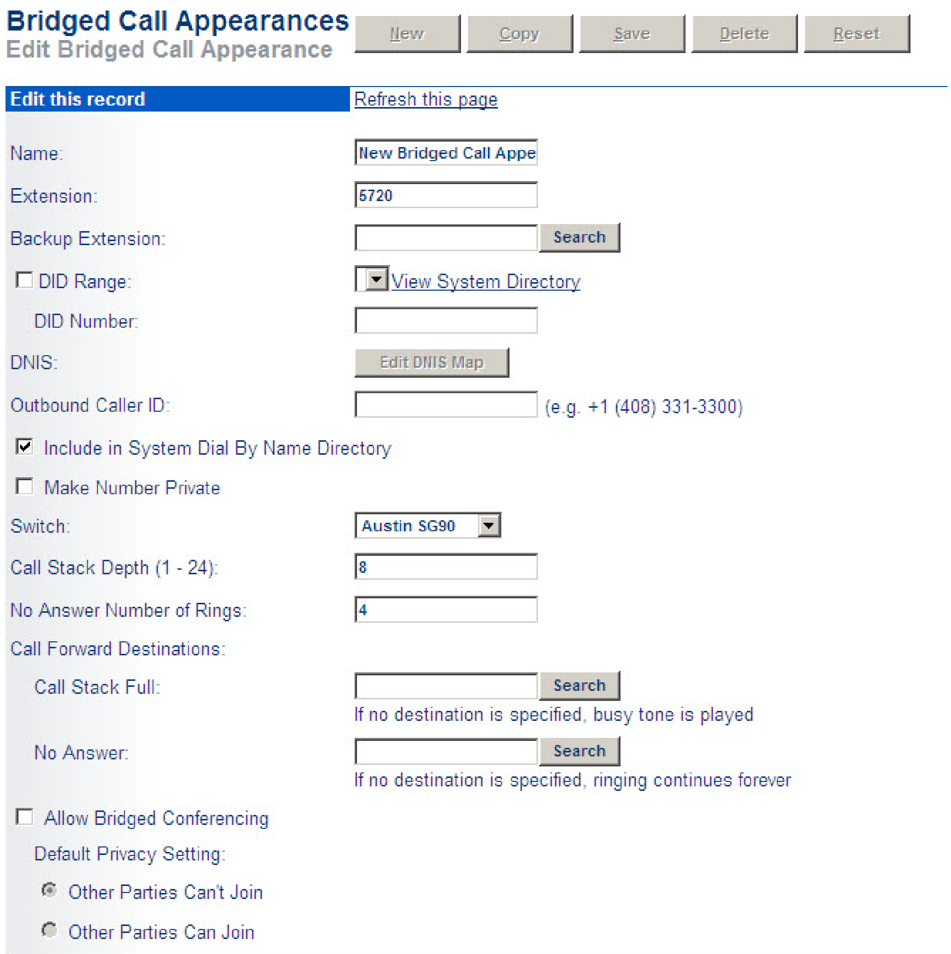


Figure 9-7 Edit Bridged Call Appearance Page

Step 4 Set the parameters for the BCA profile as described below:

- **Name:** This field specifies the label by which other Director panels and ShoreTel devices identify the BCA.
- **Extension:** This field specifies the extension on which the BCA receives a call.
- **Backup Extension:** This field specifies the extension that receives calls for the BCA when the switch that supports the BCA is out of service.

  If another BCA serves as the Backup Extension, the two BCAs must be assigned to different switches.
- **DID**: This field specifies the DID number assigned to the BCA.

  See to "DID Ranges" on page 164 for additional information about DID numbers and ranges.
- **DNIS**: This field specifies the DNIS information sent on outbound BCA calls.
- **Outbound Caller ID:** This field specifies the Caller ID number that the system sends on an outbound BCA call.

- **Include in System Dial by Name Directory:** When this parameter is enabled, the BCA extension appears on a ShoreTel IP Phone display when the user presses the Directory button.
- **Make Number Private:** This parameter designates the BCA extension as Private. The Private setting is described in "Private Numbers" on page 409.
- **Switch:** This field specifies the ShoreTel Voice Switch to which the BCA is assigned.
- **Call Stack Depth:** This field specifies the number of calls that the BCA can simultaneously handle.
- **No Answer Number of Rings:** This field specifies the No Answer condition for a call to a BCA.
- **Call Forward Destination:** These fields specify the number that should receive inbound BCA calls that are not answered in either of the following situations:
  - **Call Stack Full:** This number receives calls that are unanswered when the BCA already has the number of calls specified by the call stack depth setting.
  - **No Answer:** This number receives calls that are unanswered after the number of rings specified by the No Answer Number of Rings setting.

Step 5 Click **Save** to store your changes.

9.3.3 Configuring Answer Options on an IP Phone

This section describes how to configure BCA buttons. User answer BCA calls by pressing the green-flashing button. BCA button are buttons configured to answer calls at a specific stack position of a BCA extension.

NOTE IP Phone buttons can also be configured so that a BCA call is answered when the user lifts the handset or presses either the speaker or headset button.

Do the following to configure BCA buttons:

Step 1 Launch ShoreWare Director.

Step 2 Select **Administration > Users > Individual Users** from the Director menu. The Individual Users page appears.

Step 3 Select the user you want to configure with BCA capability. The Edit User page appears.

Step 4 Click the **Personal Options** tab.

Step 5 Click **Program IP Phone Buttons**. The Edit Custom Keys page appears as shown in Figure 9-8.

NOTE You can also click the Copy button which allows you to use the phone button profile of another user to configure the current user.

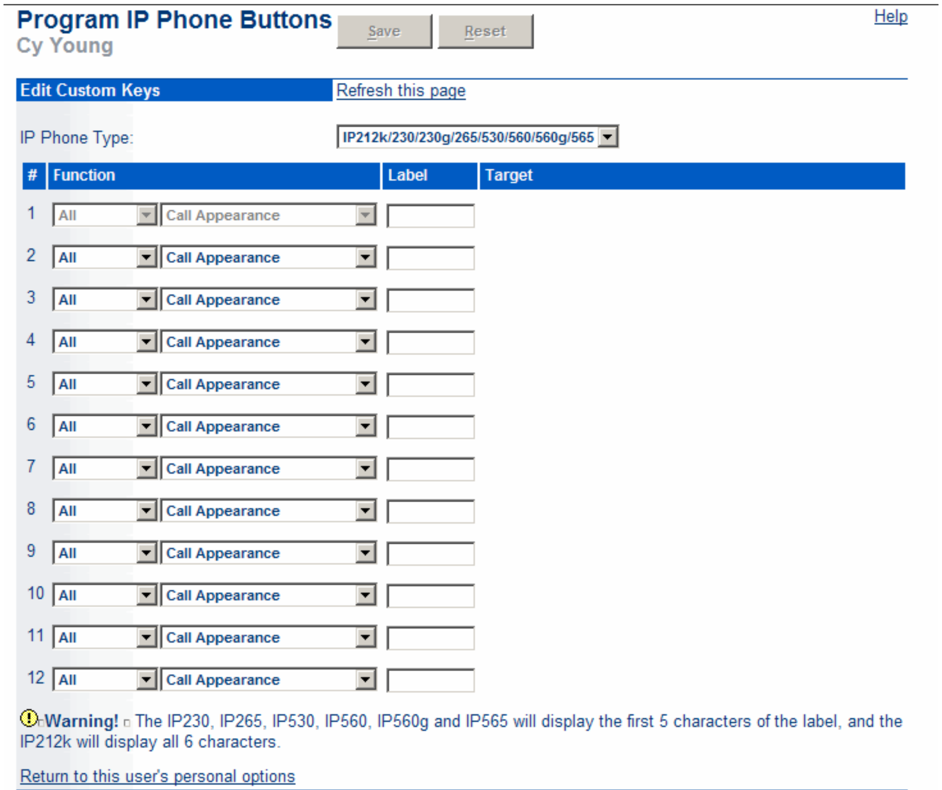


Figure 9-8 Edit Custom Keys Page

Step 6 In the IP Phone Type field, select the phone device that the user will use to answer calls.

Step 7 In the first function field, select **All** or **Telephony**.

Step 8 In the second function field, select **Bridged Call Appearance**. The Bridged Call Appearance options appears in the Target pane as shown in Figure 9-9.

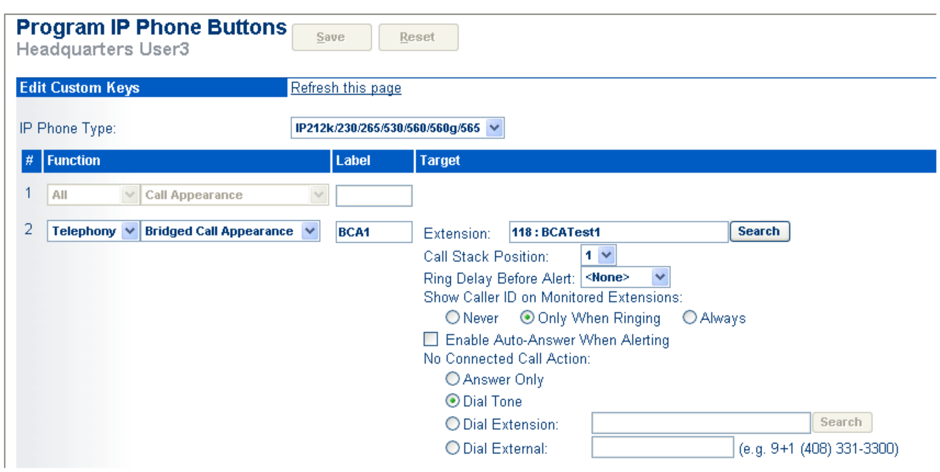


Figure 9-9 BCA Parameters in Program IP Phone Button Page

Step 9 Set the BCA IP phone button parameters as described below:

- **Extension:** This parameter identifies the BCA to which the button responds.
- **Call Stack Position:** This parameter specifies the individual calls to the BCA extension that the IP Phone Button can access.
- **Ring Delay Before Alert:** This parameter specifies the number of *inaudible* rings for an inbound call on the IP Phone before the ringing on the phone becomes audible. Valid settings include:
  - **None:** No delay—starting with the first ring, the phone rings audibly.
  - **1**, **2**, **3**, or **4**: The phone begins playing the audio alert after this number of rings.
  - **Don't Ring:** The phone never plays the audio alert.
- **Show Caller ID on Monitored Extensions:** This option specifies when the caller ID for the inbound call appears on the extension. Choices are Never, Only When Ringing, and Always. The Always choice means the caller ID remains during the conversation.
- **Enable Auto-Answer when Alerting:** When this option is selected, the Inbound BCA Call is answered when the user either presses the IP Phone button, takes the handset off-hook, presses the speaker button, presses the headset button, hook-flashes, or presses an unused Call Appearance button.

  When this option is not selected, the inbound BCA call is answered when the user presses the IP Phone button programmed for the BCA or presses the Answer button.
- **No Connected Call Action:** These radio buttons programs the phone's ringdown behavior. Refer to "Director Configuration Pages" on page 304 for more information about configuring Ringdown.
  - **Answer Only:** Select this option to disable ringdown
  - **Dial Tone:** Select this option to configure the phone as the recipient on a ringdown circuit.
  - **Dial Extension:** Select this option to configure the button as the initiating end of a ringdown circuit when the recipient is an IP phone on the ShoreTel network.
  - **Dial External:** Select this option to configure the button as the calling end of a ringdown circuit when the recipient is a device that is not on the ShoreTel network.

Step 10 Click **Save**.

9.4 Bridged Call Appearance Conferencing

This section describes BCA conferencing. BCA conferencing is available to regular BCA users and SCA users (and their assistants).

Bridged Call Appearances are set up to be private by default, so a BCA or SCA user with a call in progress cannot be joined by other BCA users on the same extension. However, the default setting can be changed to allow others to join, and an override on the phone lets the owner of the call lock or unlock the conference regardless of the default.

When a call is made to the BCA line, the flashing orange BCA button turns green when the user answers the BCA call. Other BCA users see either of the following on this line:

- A solid orange LED if conferencing is allowed by default: If the button is orange, the BCA user can press the button to join the BCA call in progress.

- A solid red LED if conferencing is disallowed by default: If the button is red, users cannot join the active BCA call unless the owner of the call presses the Unlock button on his or her phone.

Figure 9-10 shows an outside call to a BCA line.

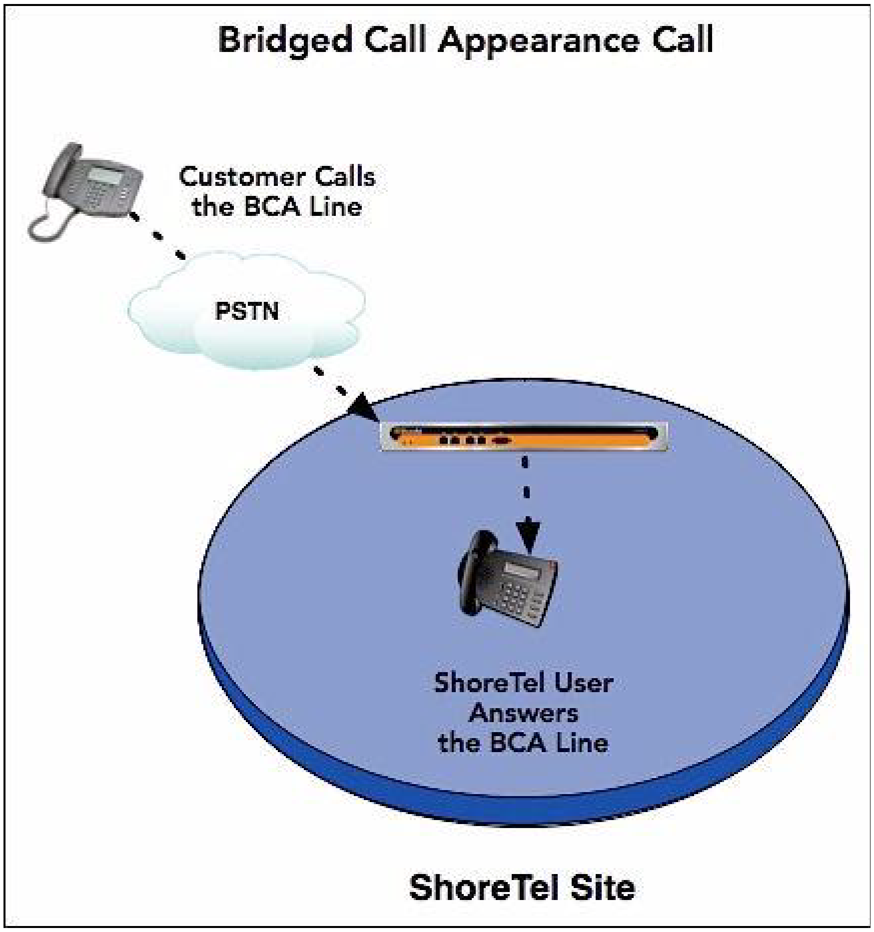


Figure 9-10 Call to a Bridged Call Appearance

With permission, a BCA user can join the active BCA call by pressing the orange BCA button on the phone. Figure 9-11 shows that other BCA users have joined a BCA call. Note that in Figure 9-11, the caller directly connects to the original BCA users' ShoreTEL IP Phone. The phone of each additional BCA user is transferred to a ShoreTel Voice Switch with available Make Me Conference ports that directly connects each additional BCA user.

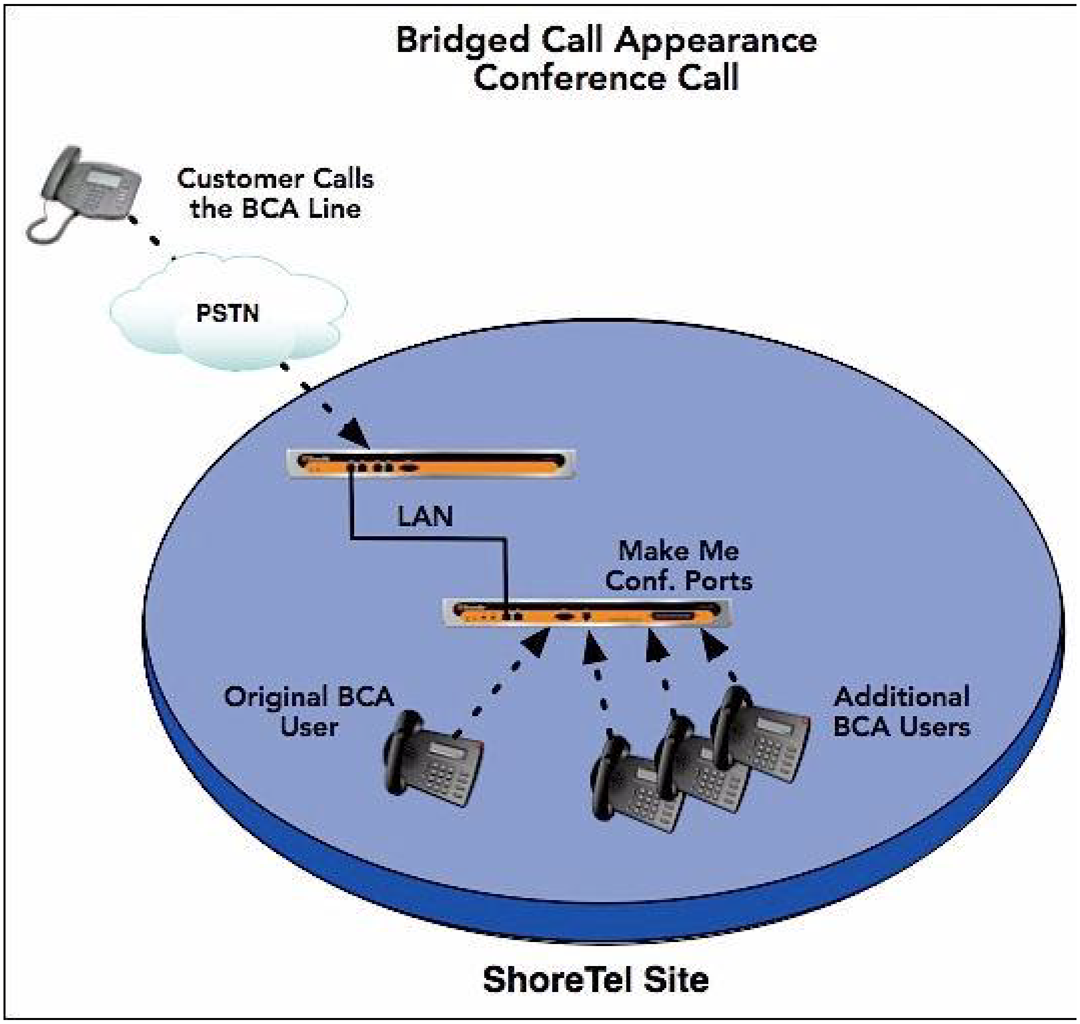


Figure 9-11 Bridged Call Appearance - Conference Call

9.4.1 Answering and Joining the BCA Call

When a call comes into an BCA line, the color of the BCA button indicates if the BCA call is ringing, has been answered, is private, or allows conferencing. When the BCA line is ringing, the BCA button programmed on each ShoreTel IP Phone shows a flashing green color. The first user that picks up the line sees a solid green button labeled for the BCA. Other users of the BCA see either an orange or red button. If the button is orange, press the button to attempt to join the call. If enough Make Me Conference ports are available and the maximum number of allowed conferenced parties has not been reached, the user will be added to the active bridged call. If the BCA button is red and the user presses the button, an error message on the phone display and will not be able to join the conference call.

9.4.2 Setting Up Conferencing Ability for BCA and SCA Users

The BCA conference parameters for either the SCA or regular BCA users are configured in the lower-left corner of the Edit Bridged Call Appearance window.

For the selected BCA user, the entire window is active. For an SCA user, only the conference area of the BCA edit window is activated. The reason is that BCA parameters for the SCA user are inherited from the SCA account. Therefore, the rest of the BCA configuration window remains inactive (grayed out).

9.4.2.1 Specifying the Settings for a Regular BCA User

This section describes how to enable conferencing for a regular BCA user. The configuration consists of the enable, default privacy setting, and the enable of a tone that sounds when another party joins the conference.

To override the default privacy setting for the current BCA call, press the Lock/Unlock softkey on the ShoreTel IP Phone (with the green LED). The text above the softkey describes the action to be applied to the active call. (The softkey text label is a toggle between Lock and Unlock.) For example, to make the call private, the user with the call should press the button labeled “Lock.” To make the call available for conferencing, the user presses when the button shows “Unlock.”

To configure a BCA to have the conferencing:

NOTE See "Configuring Answer Options on an IP Phone" on page 248 for more information on assigning the BCA to IP phone buttons.

Step 1 Log into ShoreTel Director.

Step 2 Click **Administration > Call Control > Bridged Call Appearances**.The Bridged Call Appearance List page shown in Figure 9-12 appears.

Figure 9-12 Bridge Call Appearances List Page

Step 3 Click the **New** button or click on an existing Bridged Call Appearance profile. The Edit Bridge Call Appearances page shown in Figure 9-13 appears.

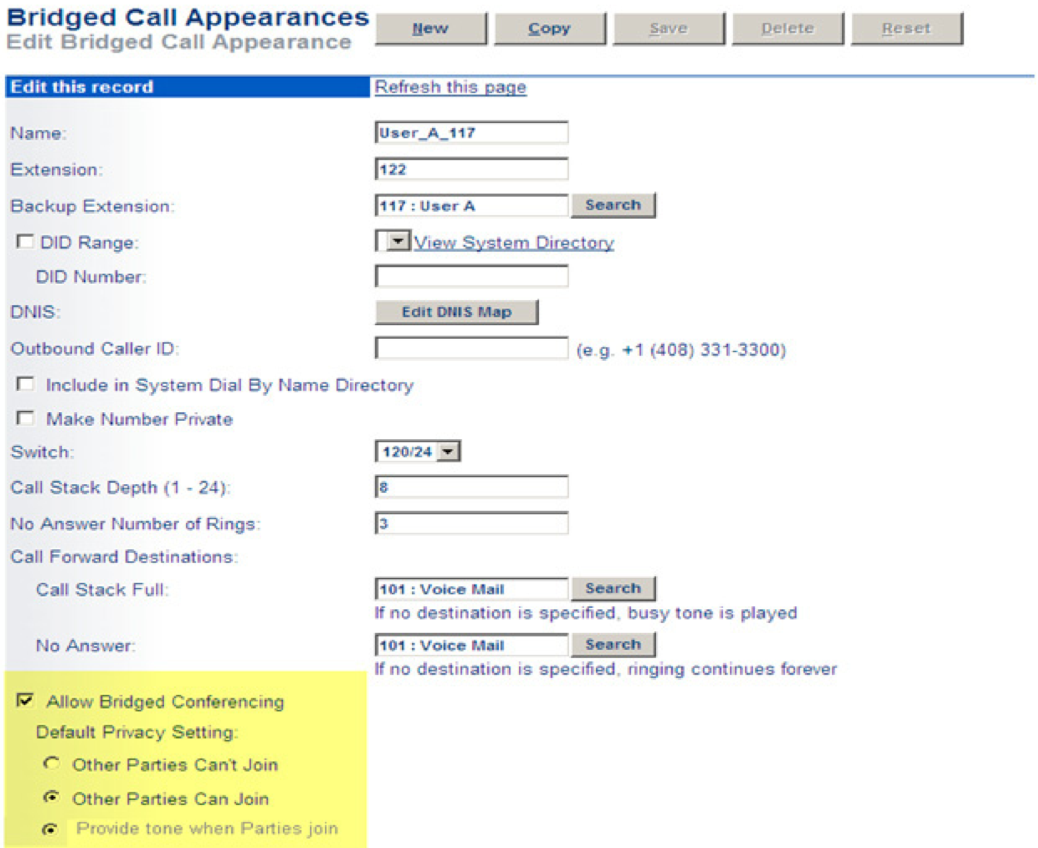


Figure 9-13 Edit Bridged Call Appearance

Step 4 Click the **Allow Bridged Conferencing** check box to enable the BCA conference facility for an SCA user or regular BCA users.

Step 5 In the Default Privacy Setting section, do one of the following:

- Select the **Other Parties Can't Join** radio button to designate that privacy is initially enabled for active BCA calls. Other users of the BCA are restricted from joining the active call. This is the default behavior.
- Select the **Other Parties Can Join** radio button to designate that the privacy feature is not initially enabled for active BCA calls. Other users of the BCA are allowed to join the active call.

Step 6 Check the **Provide Tone When Parties Join** check box to have a tone sound when a party joins the BCA conference.

Step 7 Click **Save**.

Step 8 Assign the BCA to the appropriate ShoreTel IP Phones. See "Configuring Answer Options on an IP Phone" on page 248 for more information on assigning the BCA to IP phone buttons.

Step 9 Configure the appropriate number of Make Me Conference ports on a ShoreTel Voice Switch that is available to the site. Refer to *Switch Parameters* for more information on configuring Make Me Conference ports.

9.4.2.2 Enabling the BCA Conference Ability for an SCA User

The Edit Bridge Conference Appearance page lets you:

- Enable the executive's conference ability
- Set the privacy default (the default for permitting or locking out other users)
- Enable the join-in tone

The SCA user always owns the conference call and can decide when other BCA users are admitted to the conference. An SCA user can override the default privacy setting by toggling the Lock/Unlock sofkey.

To configure BCA conferencing for an SCA user:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Call Control > Bridge Call Appearances**. The Bridged Called Appearances List page appears. Figure 9-14 shows a Bridged Called Appearances List that includes SCA users.

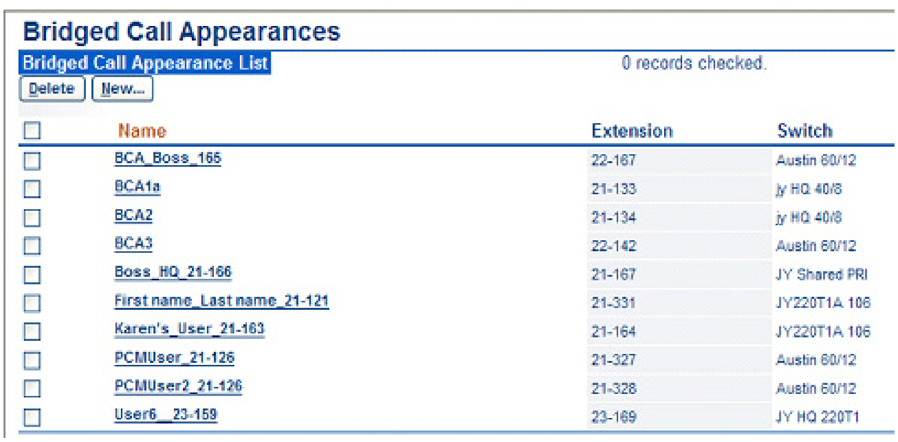


Figure 9-14 Bridge Call Appearance Page

SCA users are differentiated in the list using the format <*first name*>_<*last name*>_<*extension*>, where

- *first name* represents the first name of the user.
- *last name* represents the last name of the user.
- *extension* represents the user's extension (for SCA or regular BCA user).

Step 3 Click on the SCA user name in the Name column. The Edit Bridged Call Appearance page in Figure 9-13 on page 254 appears.

Step 4 Check the **Allow Bridged Conferencing** check box. The default privacy choice and the enable for the join tone become activated.

Step 5 In the Default Privacy Setting section, do one of the following:

- Select the **Other Parties Can't Join** radio button to designate that privacy is initially enabled for active BCA calls. Other users of the BCA are restricted from joining the active call. This is the default behavior.
- Select the **Other Parties Can Join** radio button to designate that the privacy feature is not initially enabled for active BCA calls. Other users of the BCA are allowed to join the active call.

Step 6 Check the **Provide Tone When Parties Join** check box to have a tone sound when a party joins the BCA conference.

Step 7 Click **Save**.

9.5 Shared Call Appearance

The Shared Call Appearance (SCA) feature provides call appearances that are shared between an executive (the SCA user) and an assistant (configured as a regular BCA user). The assistant can monitor the SCA user's call appearances to facilitate call handling and conferencing needs. With SCA, an assistant can help an executive with his or her communication needs by making or answering calls on behalf of the executive and by setting up phone conferences. The assistant is not restricted to supporting an executive and can receive other telephony capabilities through configuration in the ShoreTel Director. (Configuration has the flexibility to support multiple executives and assistants for different call-handling arrangements.)

For telephone conferences, the SCA user and assistant have the following:

- BCA conferencing
- Blind conferencing
- Regular conferencing abilities that all IP phone users have

BCA conferencing lets an assistant or executive set up conference calls so that, when the executive is ready, he or she can enter the conference by pressing the SCA button on the IP phone. The assistant can stay in the conference, leave the conference, or be locked out of the conference by the executive.

SCA relies on BCA as an underlying technology to support its functionality. All IP phone models except IP210, IP115, and IP110 support SCA. SIP and analog phones do not support the SCA.

9.5.1 SCA Feature Components

This section describes the main feature components of SCA. It divides components into different non-conference and conference-related areas.

9.5.1.1 Associated Bridge Call Appearance

An Associated Bridged Call Appearance (aBCA) is a bridged call appearance that is associated with an executive extension. Associated BCAs differ from other BCAs as follows:

- An aBCA can be specified in the Edit User panel rather than the Bridged Call Appearance Director panel.
- aBCAs are created when a ShoreTel extension is converted to a executive extension.

9.5.1.2 Non-conference Functionality

When a regular user is enabled for SCA, the system automatically creates an associated BCA (aBCA) gives it an aBCA extension number.

Nearly all the BCA parameters that could be selected for the regular BCA user are fixed. Only the label for each SCA button can be specified.

The settings for SCA IP programmable buttons are fixed at the following values:

- Ring Delay before Alert: None
- Show Caller ID on Monitored Extensions: Always
- Button push actions default: (unused)
- No Connection Call Action: Dial tone
- Call Stack Position: Automatically ordered (no manual ordering allowed)

The SCA call stack positions are automatically set and not manually configurable in ShoreTel Director. However, call stack positions are automatically reordered if a button is specified to be other than an SCA. The SCA buttons are reconfigured around the new button type.

NOTE For a button box, the system does not auto-shift call stack positions.

When a regular user is enabled for SCA, each regular call appearance converts to an SCA. Standard call appearances do not exist for the SCA user, and no SCA button can be converted back to a regular call appearance unless the SCA configuration is removed by disabling SCA.

9.5.1.3 BCA Conference Parameters

The BCA conference parameters are configured in the Call Control –> Edit Bridged Call Appearances window for a specific BCA or aBCA. For a description of the conferencing setup for an SCA user, see "Enabling the BCA Conference Ability for an SCA User" on page 255. The conference parameters are:

- The enable for bridged conferencing
- The Default Privacy setting—an enable for letting other BCA users join the conference (can be overridden by the executive user on a per call basis)
- A tone that sounds when a party joins a conference

9.5.2 Enabling a User for SCA

This section first introduces the program flow for specifying an SCA user configuration and then gives the steps for this task. The other, standard configuration tasks for a user are in the Configuring Users chapter.

9.5.2.1 Program Flow for Configuring SCA Users and BCA Conferencing

This section outlines a sequence of steps that a system administrator might follow to set up a new SCA user and assistant and enable BCA conferencing. The purpose of this outline is to promote smoother execution of the configuration steps. Readers who are already very familiar with BCA and SCA can go directly to the steps for specifying the accounts for SCA users and assistants as well as BCA conferencing for the executive.

In general, the configuration flow for creating an SCA user is as follows:

- A system administrator navigates to Users–>Individual Users, selects a site in the drop-down scroll box labeled “Add new user at site:” in the upper left corner, and clicks the GO button next to that scroll box.
- After configuring and saving the user account basics—thus creating a regular user account—the system administrator enables Shared Call Appearances in the lower half of the Edit User (General) window. For a new user, the basic user parameters must be saved before Director activates the Shared Call Appearances enable. This requirement is also pointed out in the pertinent configuration step.
- When Shared Call Appearances is enabled, the system automatically creates and numbers an *associated* BCA (aBCA) for the SCA.
- The system automatically generates the aBCA extension number and displays it in the Associated BCA box within the Edit User window. The system administrator can change the generated number, for example, according to network planning guidelines that mandate such numbers exist in a certain range (for resource management purposes).

  NOTE Once saved, an SCA user’s aBCA extension cannot be changed unless Shared Call Appearances is disabled for the user.

- The aBCA is deleted if the system administrator reverts the SCA user back to a normal user by clearing the user’s Shared Call Appearances checkbox in the Edit User panel.

  NOTE Before the Share Call Appearances check box can be cleared, certain programmed items must be cleared. The system administrator must clear:

  — All of the assistant’s IP buttons that monitor the executive’s aBCAs

  — The assistant’s Communicator toolbar programming for monitoring the executive’s aBCA (if present)

In summary, implementation tasks for a new SCA user and regular BCA user (assistant) have the following sequence:

1. Create the new regular user account that is intended for the SCA user.
2. Convert the new user to an SCA (executive) user by enabling SCA.
3. Configure the IP phone buttons that the SCA user’s account calls for.
4. In the Bridged Call Appearances window, enable conference ability, select the default privacy setting, and enable the join-tone for the executive. (See "Setting Up Conferencing Ability for BCA and SCA Users" on page 253.)
5. Create a new regular BCA user to have an assistant’s phone setup: configuration includes the executive’s extensions that the assistant monitors. For details, see "Creating an Assistant Account" on page 260.

9.5.2.2 Creating a New Executive User

The configuration tasks in this section apply to a new executive (SCA) user and a new administrator (regular BCA) in a subsequent section. Other types of steps for configuring a new user apply to the executive and administrator accounts, but they are largely omitted here. Refer to the chapter on creating Users for all the details on user accounts.

NOTE If the executive and assistant need to manage SCA calls in Communicator, the Access License for each of these users must be Operator.

To create a user with the SCA user:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Users > Individual Users**.

Step 3 In the “Add new user at site” field, select the site for the new user, then click **Go**. The Users-Edit Users appears. This window shows some defaults that are specific to the site and the server.

Step 4 In the First Name and Last Name fields, type a first and last of the new user. (As a subsequent step shows, Director displays first, last name, and extension number in a specific format.)

Step 5 In the Access License field, select **Professiona**l or **Operator**. ShoreTel Communicator requires the Operator license to display the BCA Window.

Step 6 In the Primary Phone Port section, click the IP Phones radio button and select a specific phone in the drop-down list. (If an IP phone model of sufficient capability is not recognizable in the list of MAC addresses, the right phone can be determined by matching the IP phone model number to a MAC address in Administration > IP Phones > Individual IP Phone.)

NOTE Before enabling SCA for a new user, the system administrator must do an intermediate Save after entering the basic user details in the upper part of the Edit Users window. The Enable SCA checkbox becomes active only after this intermediate Save.

Step 7 Check the **Shared Call Appearances** checkbox.

Step 8 In the Associated BCA, enter a number for the extension.

The next available extension number is automatically generated when Shared Call Appearance is enabled. The number can be changed. If the system administrator has an organized scheme for extension numbers, manual entry is a logical choice. A manually entered alternative could be valuable if an auto-generated number is outside the number management scheme and, therefore, possibly confusing.

NOTE A number must be deemed acceptable before the Save button is clicked to save the SCA setting. After the SCA enable has been saved, the Associated BCA value cannot be changed except by deleting the SCA (by unchecking the SCA box) and then starting the SCA configuration over.

Step 9 (Optional: This step can be skipped if the auto-fill value is acceptable.) Type an aBCA number in the Associated BCA field.

Step 10 Click **Save** if the aBCA number is acceptable.

Step 11 Click the **Personal Options** tab.

Step 12 In the Current Call Stack Size field, type a value for the depth of the call stack field.

This entry is the number of SCA buttons that the executive has. The number is subsequently reflected in the window for programming IP phone buttons (the Program IP Phone Buttons window). The system default is 8, but we recommend some planning for resource usage. For example, some executives might need only two or three while others might need eight or more, and planning should have been completed for button boxes and how many executives one assistant might have to support.

Step 13 Click the **Program IP Phone Buttons** link. The Program IP Phone Button page for the current user appears as shown in Figure 9-15.

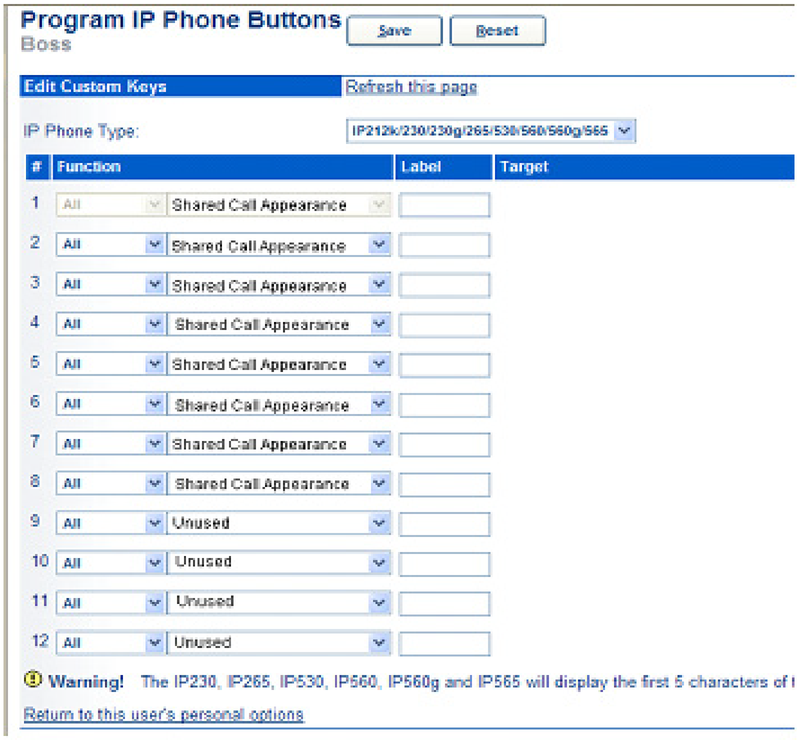


Figure 9-15 Programming the IP Phone Buttons for SCA

Step 14 In the IP Phone Type field, select type of IP phone the user uses.

Step 15 In the Label field for each SCA button, type a name. All other BCA parameters are fixed for an SCA button. Non-SCA buttons can be programmed for other functions, such as speed dial, hotline, and so on.

Step 16 Click **Save** when all necessary parameters are configured.

NOTE Clearing the Shared Call Appearances checkbox returns the executive extension to a normal extension and deletes the extension's aBCA associated.

The executive account has been created, but conferencing and default privacy settings for the SCA user still need setting up. To configure the conference-related details for the SCA user, refer to "Setting Up Conferencing Ability for BCA and SCA Users" on page 253.

9.5.3 Creating an Assistant Account

The steps in this section focus on configuring a regular BCA user to monitor the executive's call appearances. The description omits many common details for configuring a regular user. For the standard user details, see the chapter Configuring Users.

NOTE Making all the executive's call appearances visible to the assistant is not required: If an executive wants one or more lines to be invisible to the assistant, the administrator omits the requested number of hidden lines from the assistant's configuration.

After creating the assistant as a regular BCA user in Edit Users, the system administrator proceeds with configuring the assistant to monitor the executive's phones, as follows:

Step 1 Save the regular user parameters in the Edit User > General tab.

Step 2 Select the **Personal Options** tab.

Step 3 Click the Program IP Phone Buttons link. The Program IP Phone Buttons page appears.

Step 4 In the IP Phone Type field, select the type of IP phone the assistent uses.

Step 5 In the first function field, select **All** or **Telephony**.

Step 6 In the second function field, select **Bridged Call Appearance**. The Bridged Call Appearance options appears in the Target pane.

BCA parameters for assistant-monitoring of an executive's call appearances are available, but the executive's aBCA first must be *selected* after a search for it.

Step 7 Click the **Search** button. The Dialing Numbers--Webpage Dialog shown in Figure 9-16 appears.

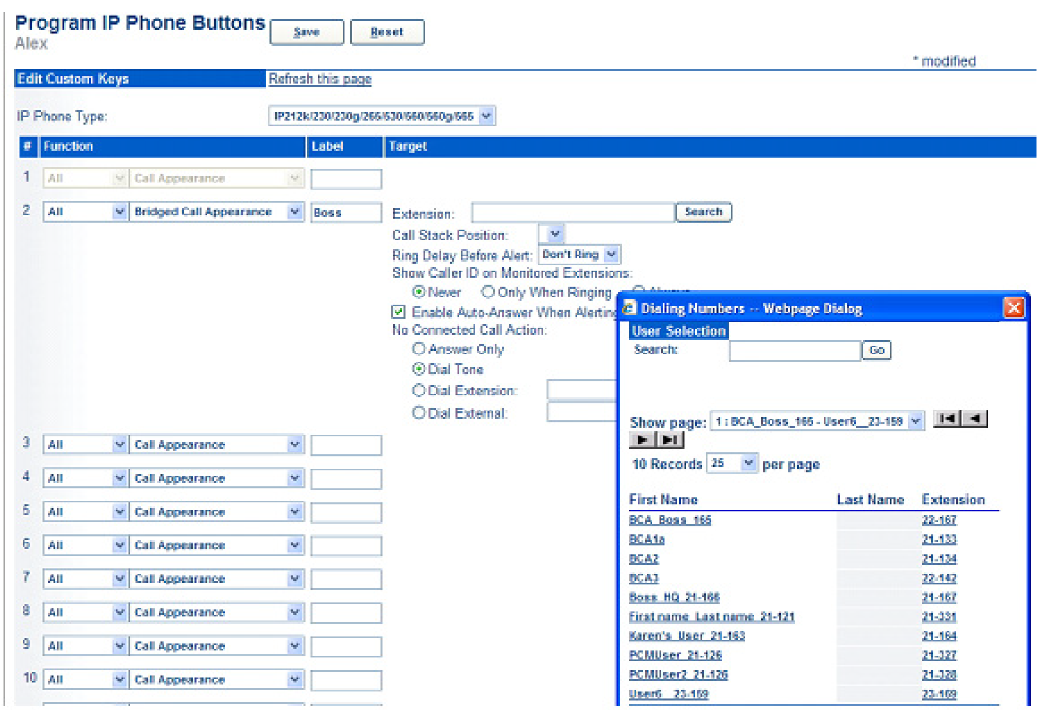


Figure 9-16 Searching for the SCA User for the Assistant to Monitor

Step 8 Select an aBCA with whom to associate the assistent. aBCAs are listed using the format: *<first name>_<last name>_<extension>*.

Step 9 Configure other BCA parameters as needed. See "Configuring Answer Options on an IP Phone" on page 248 for more information about the BCA parameters.

> **NOTE** The No Answer Number of Rings and Call Forward Destination parameters reflect the initial values of these parameters. However, the real-time state of these parameters can change, based on the activity of the user. For a regular BCA user, these parameters are editable in the BCA window in Director. In contrast, for an SCA user (tied to an aBCA), these parameters are grayed out in the window because the SCA user inherits the parameters when the system creates the aBCA. Thereafter, if the SCA user is changing these parameters in real time, the changes are actually inherited in the aBCA, but the BCA page in Director is not updated to reflect the changes. Put another way, ShoreTel Director continues to reflect the initial value of these two SCA parameters.

Step 10 Configure other call appearances as needed for this executive (or for other executives after searching the same pop-up window described in Step 7).

Step 11 Repeat as needed the configuration steps for each executive call appearance.

Step 12 Click the **Save** button when the configuration steps for this window are complete.

9.5.4 Usage Guidelines for SCA

This section contains different usage scenarios and suggestions.

When an executive extension is routed to a phone that does not have programmable buttons, the executive extension is handled as a normal extension. The extension behavior reverts back to that of an executive extension when extension calls are routed to a phone with programmable buttons.

9.5.4.1 Bridged Call Appearance Monitoring

The Bridged Call Appearance Monitor is a ShoreTel Communicator window that displays all Bridged Call Appearances accessible to the user's extension through all devices assigned to the user. The BCA Monitor consolidates all of a user's aBCA activity into a single panel.

The Bridged Call Appearance Monitor is available in ShoreTel Communicator only if the Access License for the user is Operator.

In Communicator, because calls are tracked as BCA calls in the BCA monitor window, the active call cell disappears when a call is put on hold, and the held call can be viewed only in the BCA monitor window.

aBCA is hidden in the phone directory list so that users do not accidently call an aBCA instead of the executive. However, as with regular BCAs, the system administrator can configure an aBCA as the destination of a trunk group or the targeted extension of a programmable button function. The decision of how to use aBCA belongs to the system administrator's discretion.

Placing an executive extension call on hold parks the call on the aBCA. Held calls on a executive extension are viewable from the Bridged Call Appearance Monitor in ShoreTel Communicator or from the IP Phone display.

9.5.4.2 Assistant Users

Assistant accounts need no special configuration for the monitoring of the executive's call appearances other than the assignment of programmable buttons for IP phones. However, for monitoring of executive call appearances in Communicator, the assistant's Access License must be Operator.

9.5.4.3 Hotline

A typical SCA setup includes a hotline circuit between an executive and assistant. They use a hotline circuit to communicate requests, responses, and status of calls.

To land a call on a hotline button for intercom or speed dial, both parties must have a hotline-programmed button. In the absence of this programming, the offered call is processed as a regular call.

Hotline calls and Extension Monitor calls to an executive extension that are picked up are not bridged. Refer to the Hotline section for configuration information.

NOTE A hotline intercom call uses the intercom permissions of the user. Therefore, the rules that apply to that user's regular intercom call also apply to a hotline-intercom call.

9.5.5 Inbound and Outbound SCA Calls

This section describes the typical actions that executives and assistants take when the assistant takes a call on behalf of an executive and when the assistant places a call on behalf of the executive. In this section, blind conferencing is not used. Examples of blind conferencing are in "Blind Conferencing and the SCA User" on page 264.

9.5.5.1 Assistant Support for Inbound Calls

The following scenario describes a typical sequence of actions when an assistant takes a call on behalf of an executive.

1. The inbound call triggers a flashing orange light on the IP phone of both the assistant and the executive. If the Access License of the executive and assistant is Operator, Communicator also signals the incoming call. (The executive could just pick up the call and pre-empt the assistant's involvement with this call.)
2. The assistant answers the call on the flashing BCA button and could, for example, get the caller's name and purpose.
3. The assistant puts the call on hold.

   NOTE The executive call timer is reset if the executive puts the call on hold.
4. The assistant presses the hotline button shared with the executive.
5. The assistant tells the executive of the call in progress (on hold) and gives pertinent information about the call.
6. The executive picks up the call by pressing the flashing orange SCA button.

   NOTE Internal users who call an executive see the called party ID aBCA while the phone is ringing and the actual executive number after the call is picked up.

9.5.5.2 Assistant Support for Outbound Calls

The following scenario describes a typical sequence of actions when the assistant sets up a call for the executive.

1. The assistant accesses one of the executive's call appearances by pressing an appropriate IP Phone or ShoreTel Communicator button.
2. The assistant calls the intended recipient of the executive's call.
3. The assistant user places the called party on hold.
4. The assistant presses the hotline button to the executive.
5. The assistant tells the executive of the call in progress (on hold) and provides information about the call as needed.
6. The executive takes the call by pressing the flashing orange button that the assistant has identified.

NOTE An executive extension's redial list shows only outbound calls.

9.5.6 Blind Conferencing and the SCA User

This section illustrates blind conferencing and the SCA user in two contexts. In one situation, the assistant receives a call on behalf of the executive while the latter is already on a call. In the other context, the executive is on a call but then asks the assistant to call someone and conference the called party into the existing call.

Blind Conferencing of an Inbound Call

In this scenario:

1. The executive is currently on a call with party number one.
2. The assistant receives a call from a second party.
3. The assistant determines that the executive is on a call and wants the second party to join the executive's call.
4. The assistant hotlines the executive to say that the second party is on the line and ready to join the call.
5. The hotline call ends, and the executive is connected back to party one, and the assistant is connected back to party two.
6. The assistant initiates a conference and selects the executive's call into which party two must join.

After the conference connection is completed, parties one and two are in the same call.

Blind Conferencing of an Outbound Call

In this scenario:

1. The executive is on the phone with party one and uses the hotline to ask the assistant to bring another party into the call
2. The executive goes back on-line with party one.
3. The assistant calls party two.
4. The assistant hotlines the executive to say that party two is ready to join.

5. The assistant adds party two by initiating the blind conference and then pushing the button for the executive's active call appearance.

9.6 Silent Coach

Silent Coach is a client feature that lets a user (the initiator) intervene in another user's active call and communicate with that user (the recipient). The initiator can speak to the recipient and listen to all other call participants on the call. The recipient is the only call participant that can hear the initiator.

The right to use Silent Coach is set by the system administrator. The system administrator also specifies the users (recipients) whose calls the initiator can monitor. A Telephony Class of Service assigns Silent Coach rights. Silent Coach can be initiated through various ShoreTel IP Phone models or through ShoreTel Communicator.

NOTE Silent Coach is not available on SIP extensions or over SIP phones trunks.

9.6.1 Operational Behaviors of the Silent Coach Feature

The following are details about Silent Coach Behavior:

- Silent Coach lets the initiator switch between Silent Monitor, Barge In, and Silent Coach functions for the same call.
- Silent Coach sessions can be initiated through ShoreTel IP Phone or ShoreTel Communicator programmable buttons, ShoreTel Communicator menu options, and star code calls from other calling devices.
- The initiator of a Silent Coach session can change the session to a Silent Monitor or Barge In session. Silent Monitor sessions can be changed into a Silent Coach sessions.
- The recipient can place the original call on hold to engage in a two-way conversation with the Silent Coach initiator. At the end of this conversation, the user can resume or terminate the original call.
- Silent Coach cannot be initiated with users who are on conference calls.
- A call with an active Silent Coach session cannot be transferred or converted to a conference call.
- The recipient cannot record calls while Silent Coach is active.

NOTE The following devices do not support session transitions, coach consulting, and coach resumption.

- Analog phones
- IP110
- IP210

9.6.2 Configuring Silent Coach Permissions

Silent Coach access is controlled through Telephony CoS settings. Permissions for monitoring calls or having calls monitored are configured through the Silent Monitor /

Silent Coach option on the Telephony Class of Service Edit panel. To configure Silent Coach for a Telephony Class of Service:

Step 1 Launch ShoreWare Director.

Step 2 Click **Administrator -> Users -> Class of Service**.

In the Telephony Features Permissions section, click the feature profile that you want to configure or "Add new" to create a new feature profile. The Edit Telephony Features Permission page appears as shown in Figure 9-10.

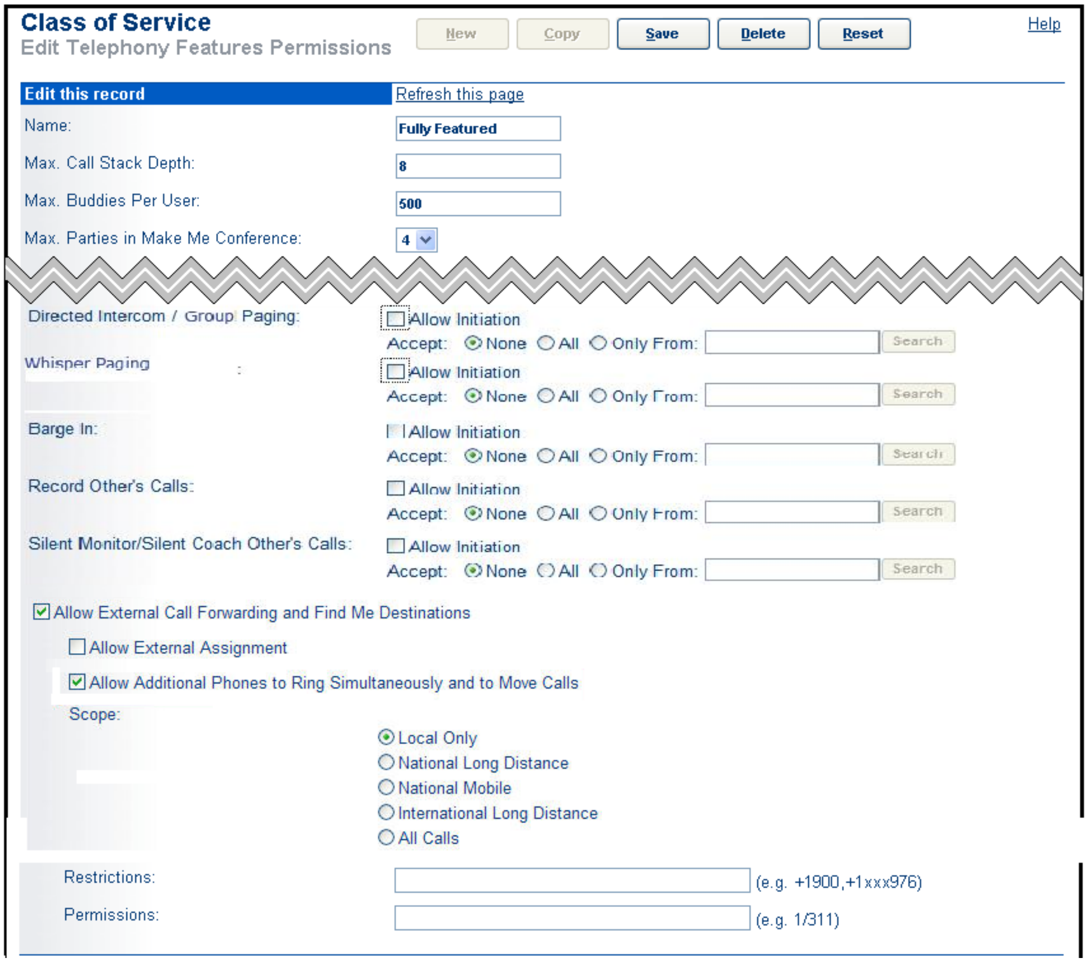


Figure 9-17 Class of Service Setting – Silent Coach

Step 3 Scroll down to the Silent Monitor / Silent Coach Other's Calls section and do the following:

- Check the **Allow Initiation** checkbox.
- In the Accept section, check the radio button of the permission you want users with this CoS to have to monitor BCA calls. The option are:
  - — **None:** Select to allow users to monitor calls.
  - — **All:** Select to allow users to monitor all calls.

— **From Only**: Select to allow users to monitor specific call participants. In the From Only field, seclect the participant for whom you want to allow monitoring.

Step 4 Click **Save**.

9.6.3 Enabling the Silent Coach Warning Tone

ShoreTel provides an option for playing a Silent Coach Warning Tone to all call participants when a Silent Coach session is initiated and if the Silent Coach Warning Tone option is enabled. The Warning Tone setting applies to all Silent Coach sessions on the system.When a user transitions between Silent Coach and Silent monitor, the warning tones start/stop are based on the silent coach or silent monitor warning tone setting.

To enable the Silent Coach Warning Tone option:

Step 1 Launch ShoreWare Director.

Step 1 Click **Administration > Call Control > Options**. The Call Control Options Edit page appears as shown in Figure 9-18.

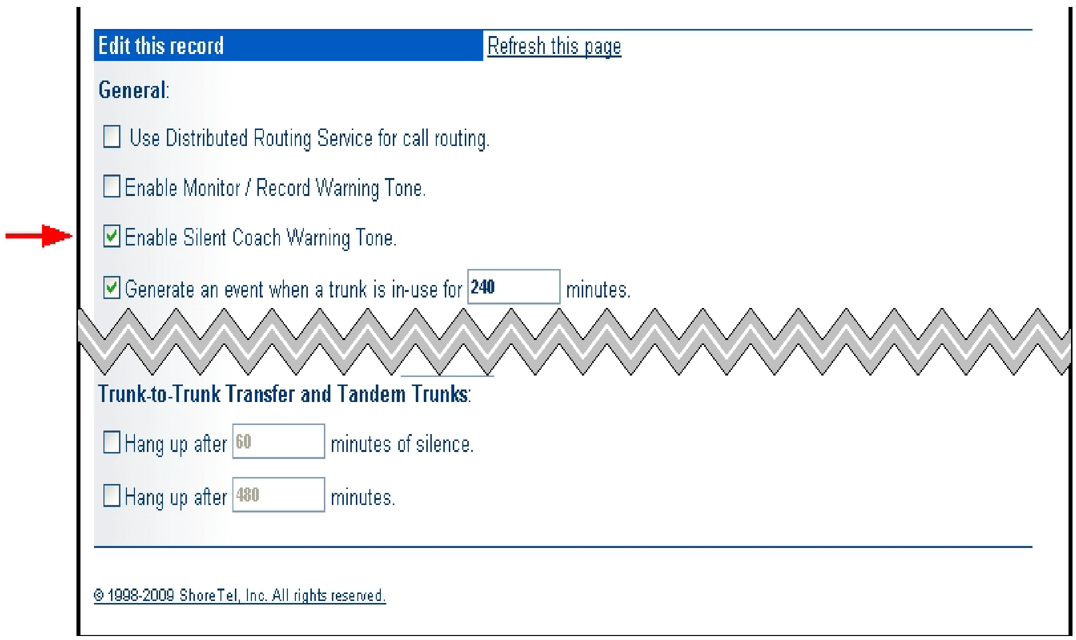


Figure 9-18 Enabling the Silent Coach Warning Tone

Step 2 Check the **Enable Silent Coach Warning Tone** check box.

9.6.4 Configuring Silent Coach buttons

ShoreTel IP Phone and ShoreTel Communicator programmable buttons can be configured to initiate a Silent Coach session with a specific user or to query the caller for a Silent Coach destination. The configuration processes for ShoreTel IP Phone and ShoreTel Communicator programmable buttons are almost identical.

To configure an IP Phone button to initiate Silent Coach:

Step 1 Launch ShoreWare Director.

Step 2 Click **Administration > Users > Individual Users**. The Individual Users page appears.

Step 3 Select the user that you want to be able to initiate the silent coach feature. The Edit User page appears.

Step 4 Click the **Personal Options** tab.

Step 5 Click the **Program IP Phone Buttons link**. The Program IP Phone Buttons page appears.

Step 6 For the IP phone button that you want the user to use, select **All** or **Telephony** in the first field.

Step 7 In the second field, select **Silent Coach**. The Target pane in the Program IP Phone Buttons page populates as shown in Figure 9-19.

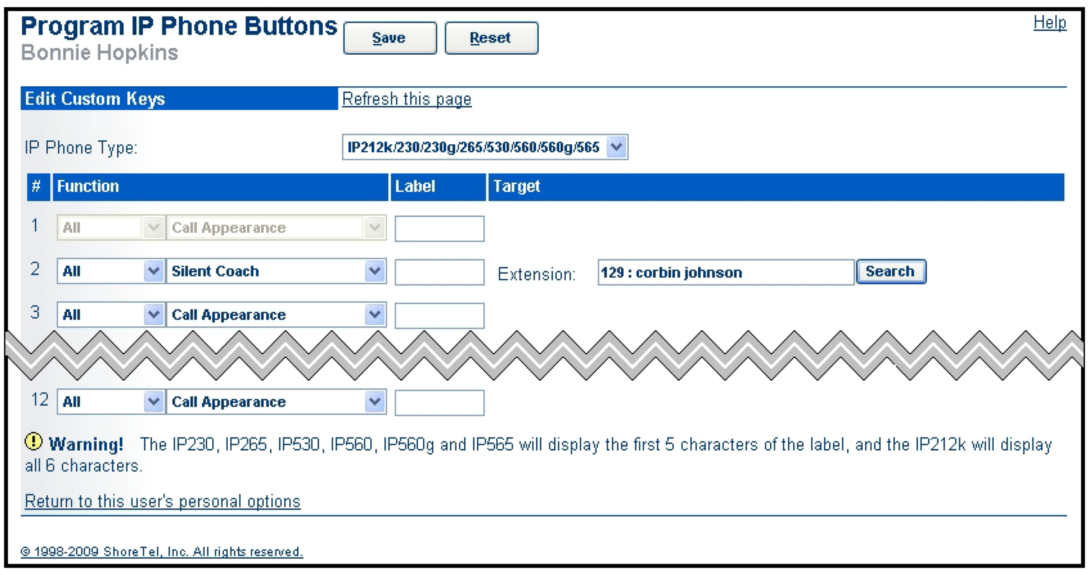


Figure 9-19 Programming IP Phone Button – Silent Coach

Step 8 Select Silent Coach from the ***Function*** drop-down menu of a selected button.

Step 9 In the Label field, enter a name for the button.

Step 10 To program the button to monitor the calls of a specific user, enter the user's extension in the Extension field.

> NOTE If this field is left blank, pressing the IP Phone button queries the initiator to enter a number in the Telephone User Interface before the session begins.

Step 11 Click **Save**.

9.6.5 Performing Silent Coach Operations

Silent Coach is initiated from the following:

- ShoreTel Communicator programmable button
- ShoreTel Communicator menu option
- ShoreTel IP Phone programmable button
- Any phone by pressing the code *22, then entering the target extension.

9.6.5.1 Initiating Silent Coach Operations with ShoreTel Communicator

ShoreTel Communicator supports two methods of initiating Silent Coach operations – from menu options and by pressing a tool bar button.

To perform a Silent Coach from a ShoreTel Communicator Menu:

Step 1 Perform one of the following:

- Click the Application Button, then select **Dial > Silent Coach.**
- Select **Dial > Silent Coach** from the main menu.
- Right click the System Tray ShoreTel icon select **Dial > Silent Coach.**

ShoreTel Communicator displays the **Silent Coach** dialog box, as shown in Figure 9-20.

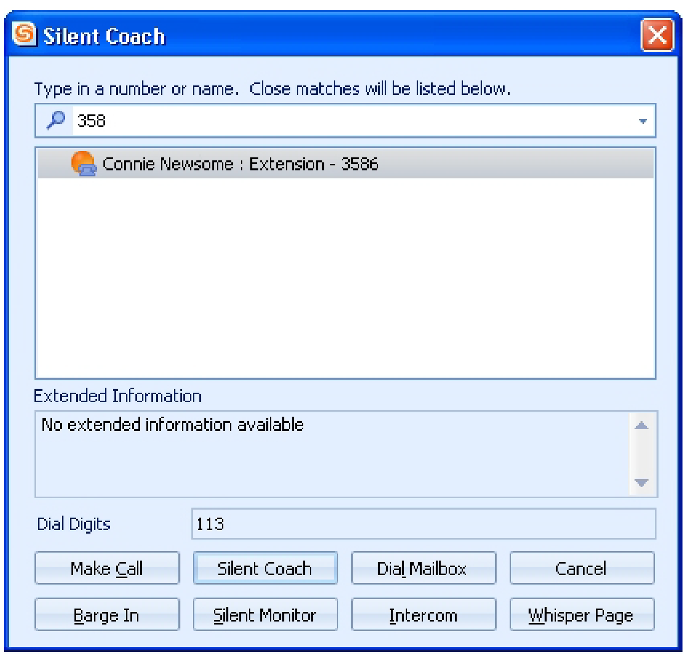


Figure 9-20 Silent Coach Dialog box

Step 2 Enter an extension or select an item from the drop-down list.

Step 3 Click the **Silent Coach** button at the bottom of the panel.

To perform a Silent Coach from a ShoreTel Communicator tool bar button:

Step 1 Press the programmable ***Silent Coach*** button, as shown in Figure 9-21.

NOTE If the button specifies a Silent Coach recipient, the system immediately initiates a Silent Coach session with that user. Skip the remaining steps.

NOTE If the button does not specify a Silent Coach recipient, ShoreTel Communicator displays the Si**lent Coach** dialog box. In this case, continue to the next step.

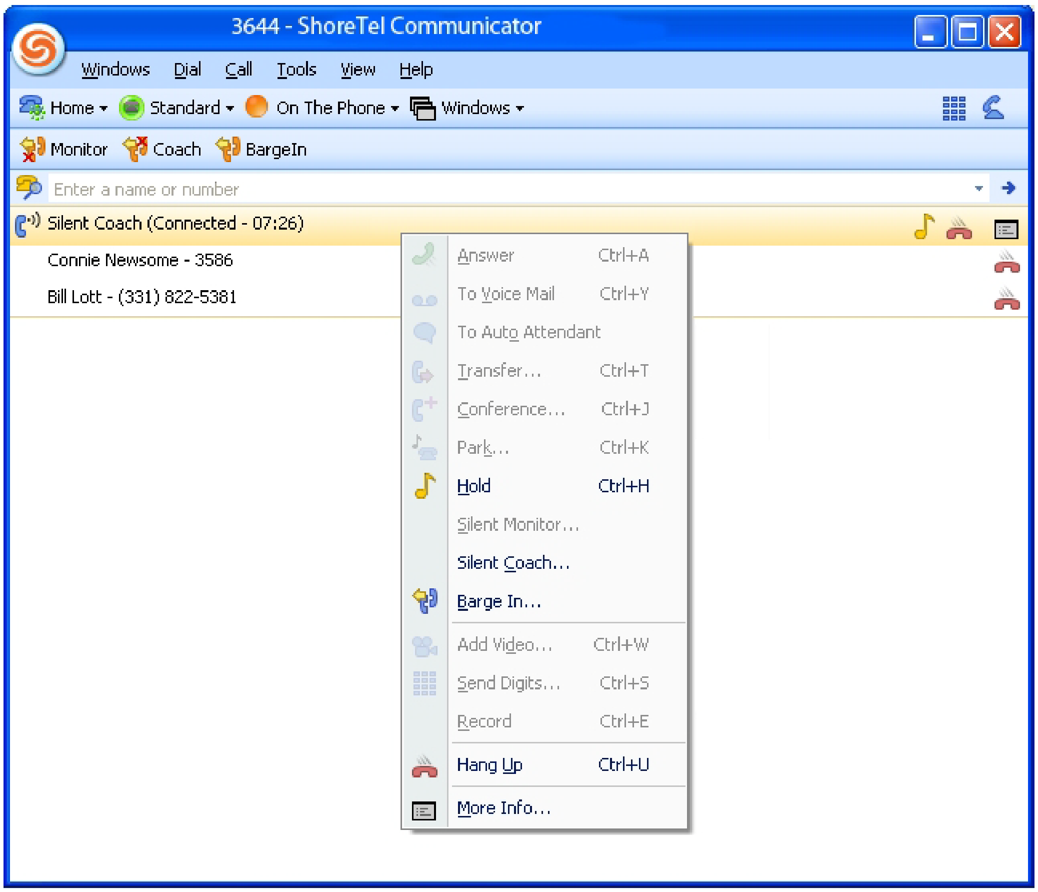


Figure 9-21 ShoreTel Communicator – Silent Coach session

Step 2 Enter an extension or select from the drop-down list.

Step 3 Click the **Silent Coach** button.

To transition a Silent Coach session into a Silent Monitor or Barge In session, perform one of the following:

- Right click the call cell and select the desired session type on the context menu.

  Figure 9-21 displays the context menu during a Silent Monitor Session.

- If available, press the Monitor or Barge In programmable button.

  Figure 9-21 displays Silent Monitor, Silent Coach, and Barge In programmable buttons.

The same process transitions a Silent Monitor session into a Silent Coach session.

9.6.5.2 Initiating Silent Coach Operations with ShoreTel IP Phones

To perform a Silent Coach from a ShoreTel IP Phone programmable button, press the Silent Coach button.

- If the button specifies a Silent Coach recipient, the system immediately initiates a Silent Coach session with that user. Disregard the remaining steps.
- If the button does not specify a Silent Coach recipient, enter the recipient's name or number in the Telephone User Interface.

ShoreTel Communicator displays the **Silent Coach** dialog box. In this case, continue to the next step.

To perform a Silent Coach from any system phone,

- Enter the *22 code, followed by the number of the target extension.

ShoreTel phones display softphone options while a Silent Monitor option is active. Figure 9-22 displays the ShoreTel IP560 Telephone User Interface for a user that initiated a Silent Coach session.

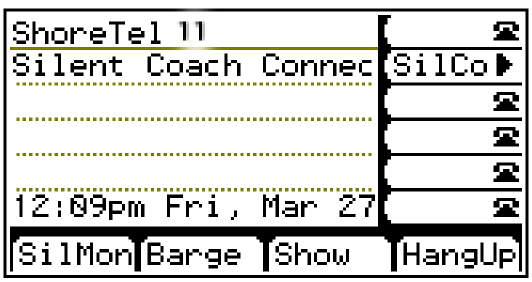


Figure 9-22 IP Phone User Interface – Silent Coach Initiator

Softkey options available to the Silent Coach initiator:

- **SilMon:** Transitions the session into a Silent Monitor session
- **Barge:** Transitions the session into a Barge In call.
- **Show:** Displays all call participants in the Telephone User Interface
- **Hangup:** Terminates the Silent Monitor session.

Softkey options available to the Silent Coach recipient:

- **Consul:** Places the active call on hold and establishes a two-way voice path between with the Silent Coach initiator.

  While the recipient consults with the initiator, soft key options include:

  — **Resume:** Restarts the recipient's original call.
  — **Show:** Displays all call participants.
  — **HangUp:** Terminates the call.

- **Show:** Displays all call participants in the Telephone User Interface.
- **Hangup:** Terminates the Silent Monitor session.

9.7 Hunt Groups

The Hunt Groups list page is shown in Figure 9-23.

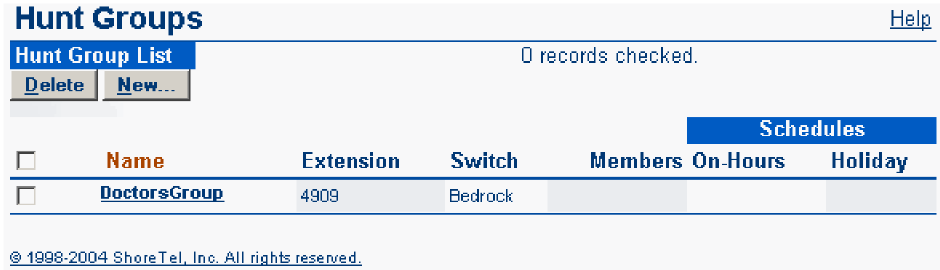


Figure 9-23 Hunt Groups List Page

Clicking on a group name shows the details for the hunt group, as shown in Figure 9-24 in the Hunt Groups edit page.

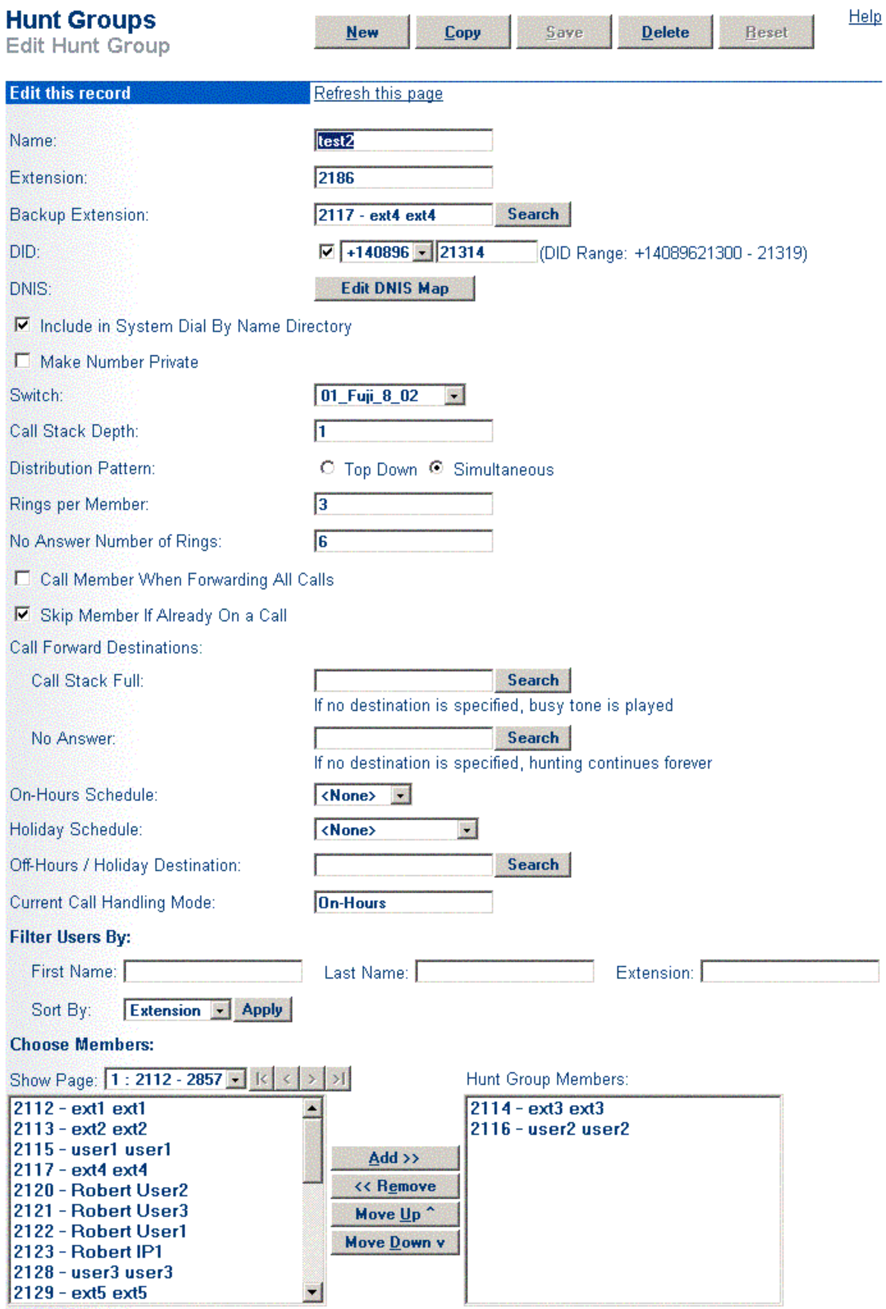


Figure 9-24 Configuring Hunt Groups

Hunt groups allow a call to be offered to a limited set of user extensions with no reporting, queuing, sophisticated schedules, log-in, log-out, or wrap-up states. Each hunt group is composed of an ordered list of no more than 16 users. A maximum of 8 hunt groups totalling no more than 16 members can be assigned to a single switch. If your requirements more complex, you should use workgroups.

Rather than being reliant on the Headquarters Server, a hunt group can be assigned to the switch closest to the agents and/or trunks associated with it. The switch controls the hunting, with no dependency upon the server. Hunt Groups have an extension number and, optionally, can also have a DID and/or DNIS number. They can be call forward extensions for users, workgroups, route points, personal assistants, site fax redirect extensions, site operator extensions, and the target for trunk groups. They are also allowed as the backup destination for workgroups and route points. This can be useful to allow some basic call handling when the workgroup server is not reachable.

The caller ID displayed for a hunt call is the external caller's ID.

A user may belong to more than one hunt group. In addition, a user assigned to a workgroup may also be assigned to hunt groups. Each call is hunted as a new call; that is, if the hunt mode is top down, each new call begins hunting from the top of the list. In this case, the person at the top of the list will get most of the calls.

Fields are defined as follows:

- **Name:** This is the name of the hunt group. Each hunt group name must be unique.

  This is a required field.

- **Extension:** This is the extension number of the hunt group. Each hunt group extension must be unique.

  This is a required field.

- **Backup Extension:** This is the backup extension of the hunt group. If the hunt group is unreachable or the switch is down, calls can be directed to this extension. A backup extension may be another hunt group, a workgroup, a route point, or a user.

  This is a required field.

- **DID:** You can assign one DID number to a hunt group.

  Check the first box to select DID. Make a selection from the drop-down list of area codes. This is an optional field.

  Refer to "DID Ranges" on page -164 for additional information about DID numbers and ranges.

- **DNIS:** The Edit DNIS Map button invokes the Select DNIS Trunk Group dialog box. This lets you select a trunk group for DNIS routing. Only trunk groups that are configured for DNIS will be presented in the dialog box. You can assign multiple DNIS numbers to a hunt group.

  DNIS is typically used to route 800-number calls to a workgroup or application.

  This is an optional field.

- **Include in System Dial By Name Directory:** This check box includes the hunt group in the auto-attendant dial-by-name directory. No name is recorded for a hunt group. When a hunt group is chosen, the extension is announced by a generic message.

  This is an optional field.

- **Make Number Private:** This check box makes the hunt group extension private. When the hunt group is private, it is not listed in the system directory and does not appear in the Communicator dialing lists. See also the "Individual Users" on page 329.

  This is an optional field.

- **Switch:** Select from the drop-down list of available switches. This is the switch that will host the hunt group and do the hunting of calls

- **Call Stack Depth**: This lets you specify the maximum number of simultaneous calls that can be "stacked" on the hunt group extension. When this number is met, additional inbound calls will be routed to the Busy Destination.

  Default value is 8. Valid entries are 1 through 16.

- **Distribution Pattern**: Click either Top Down or Simultaneous. Top Down hunts sequentially through the ordered list of group members. Simultaneous rings all group members at the same time. The first to answer is presented with the call. The default is Top Down.

  This is a required field.

- **Rings Per Member**: The default is 2 rings. All group member extensions ring with the same number of rings. If the phone is not answered, the hunt continues on to the next group member.

  This is a required field.

- **No Answer Number of Rings:** The default is 6 rings. This value determines the number of ring backs a caller will hear while the call is being hunted. Once this value is exceeded, the call is sent to the No Answer Destination.

  This is a required field.

- **Call Member When Forwarding All Calls:** Default is disabled. When enabled, even if a group member's call handling mode is Call Forward Always, the call is offered to the member.

- **Skip Member if Already on a Call**: Default is disabled. When enabled, even if a group member's call stack is not full, if the member's phone is busy or currently being offered a call, the new call is not offered to the member.

- **Busy Destination**: An alternate call destination can be specified for calls to be sent when all members of the hunt group are busy and the call stacks are full.

  This is an optional field.

- **No Answer Destination:** An alternate call destination should be specified for times when no member answers a call. The hunt will continue until the No Answer Number of Rings value is exceeded, after which callers are sent to this No Answer Destination.

- **On-Hours Schedule**: From the drop-down list, select an on-hours schedule or None. Selecting None causes all calls to be treated as if it is on-hours.

- **Holiday Schedule:** From the drop-down list, select a holiday schedule or None. Holidays are handled the same as off-hours.

- **Off-Hours/Holiday Destination**: Each hunt group can have a call forward destination for use during the off-hours state.

  This is an optional field.

- **Current Call Handling Mode:** You can configure several call handling modes for the hunt group: On-Hours, Off-Hours, or Holiday. The default is On-Hours.

  This is an informational field.

- **Choose Members:** Click a member name and then click Add to add a member to the Hunt Group. Members can be removed and re-ordered, as well. This is useful since the Hunt Group membership is an ordered list.

NOTE You can use filters to sort available members into an order that makes your selection process easiest for you.

9.7.1 Setting the Hunt Group to Busy

Users with correct permissions can set the hunt group to a busy state from the Switch Maintenance page or from the telephone user interface by using a star code (*18) followed by the hunt group extension. A confirmation prompt is played to confirm the state of the hunt group following entry of the star code. The *18 code is used to place the hunt group back into service, as well.

You can busy out the hunt group when all members are unavailable. Calls are then forwarded to the Busy Destination. After a switch reboots, the hunt group is available, by default.

Hunt Groups may also be busied out or returned to service from the Switch Maintenance page. From the Maintenance Quick Look page, select the switch that handles the hunt group, select a hunt group, and then change the busy state.

9.8 Paging Groups

As an alternative to using an in-house paging system, you can broadcast a message over a group of speakerphones with the Paging Groups feature. This feature allows a system administrator to designate groups of extensions that can be paged by dialing a single system extension and recording your message. This feature can be a cost-effective alternative for environments that do not already have an overhead paging system installed.

For environments that have an overhead paging system, Paging Groups can target your message to a select group of individuals within the organization while not exposing the message to everyone in the building, as would happen with an overhead page.

9.8.1 Adding Overhead Paging to Paging Groups

A Paging Extension (i.e. the extension which, when dialed, sends a page announcement to a site's overhead paging system) can be included in a Paging Group. By merging a Paging Extension within a Paging Group, you can broadcast a message to a select group of user extensions AND send it to the overhead paging system at the same time. Adding multiple Paging Extensions to an extension list provides the capability to simultaneously page the overhead paging system of multiple sites.

- Paging Group messages sent to an IP phone that is on-hook are announced on the speaker of that IP phone. Pages sent to an IP phone or analog phone that is already on a call are treated as a normal call.
- Call handling does not apply to paging calls.
- A maximum of 100 extensions can be paged at one time.
- After receiving a request to play a paging message, the workgroup server introduces a short pause to synchronize the audio across several phones. This pause (which can last up to 6 seconds) allows calls to all affected extensions to connect to the server, after which the server begins playing the message.

This pause synchronizes the audio paging across a group of devices, reducing the chance of a cluster of phones playing the message at different times (which could create a disconcerting echoing effect if the phones were within earshot of one another). Refer to Product Bulletin 0200 on the ShoreCare website for more information.

9.8.2 Configuration

To specify who should be paged, create an extension list of paging user numbers. For information about creating extension lists, refer to Section 10.6.4. Users in the extension list used as a paging group must belong to a user group that allows overhead paging. As an alternative to calling the paging number, group paging is available from the auto-attendant (for internal users) and can be used if this fulfills your paging needs. Group paging is not available to external callers.

Paging groups are viewed, edited, or deleted on the Call Control Paging Groups list page.

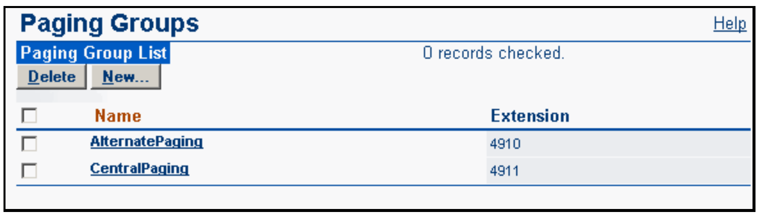


Figure 9-25 Paging Groups List Page

Select an existing group for editing or click New to add a new paging group. The Paging Groups edit page is shown in Figure 9-26.

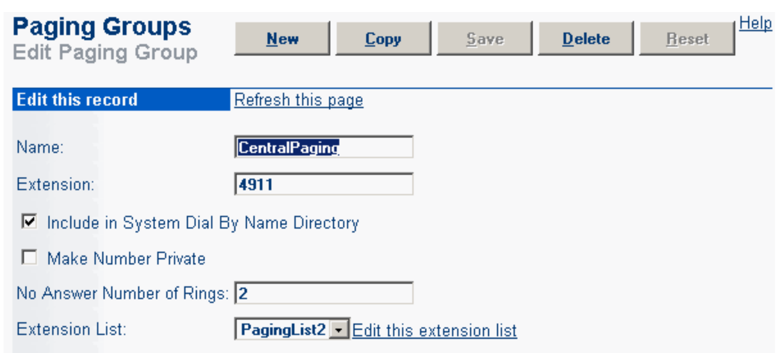


Figure 9-26 Edit Paging Group Page

The fields on the Paging Groups edit page are:

- **Name:** This is the name of the paging group. Each paging group name must be unique. This is a required field.
- **Extension:** This is the extension number of the paging group. Each paging group extension must be unique. This is a required field.
- **Include in System Dial By Name Directory:** This check box includes the paging group in the auto-attendant dial-by-name directory. No name is recorded for a paging group. When a paging group is chosen, the extension is announced by a generic message. This is an optional field.

- **Make Number Private:** This check box makes the paging group extension private. When the paging group is private, it is not listed in the system directory and does not appear in the Communicator dialing lists.

  This is an optional field.
- **No Answer Number of Rings:** The default is 2 rings. This is always used for analog phones. It is used for IP phones if the phone is busy. This is a required field.
- **Extension List:** Select the name of the extension list to be used as the paging group from the drop-down list.

9.9 Multi-site Paging Groups

The distributed nature of business often requires that business tools available to employees in the corporate headquarters also be available to remote office workers. One such business tool is a paging system. Many businesses need to have a quick way to alert employees that a customer needs assistance or that a call is waiting.

The Multi-site Paging Group feature allows employees to be paged in each remote office in an efficient manner. Multi-site Paging Groups is a ShoreTel enhancement that improves paging efficiency by allowing the audio for the page to be recorded and sent from a local ShoreTel Voicemail Server. This reduces any impact on WAN bandwidth for pages made within the Headquarters and the remote offices including any dependency on the Headquarter server.

- The Multi-site Paging Group feature allows users to pick up a phone and dial a single system extension to page a group of telephones. With Multi-site Paging Groups, the administrator can now configure local paging extensions for each site.
- The Multi-site Paging Group functionality can be implemented on the Headquarters Server and Distributed Voice Mail Servers. This feature is configured in a similar manner to the Paging Group feature implemented in previous releases.

NOTE Group paging is not available on Voice Model Switches (switches with the letter V appended to the model name—the ShoreTel Voice Switch 90V, for example).

Figure 9-27 shows a two site implementation where each site has a Paging Group. With Multi-site Paging Groups, both pages are recorded and sent by their local server with no impact on WAN bandwidth.

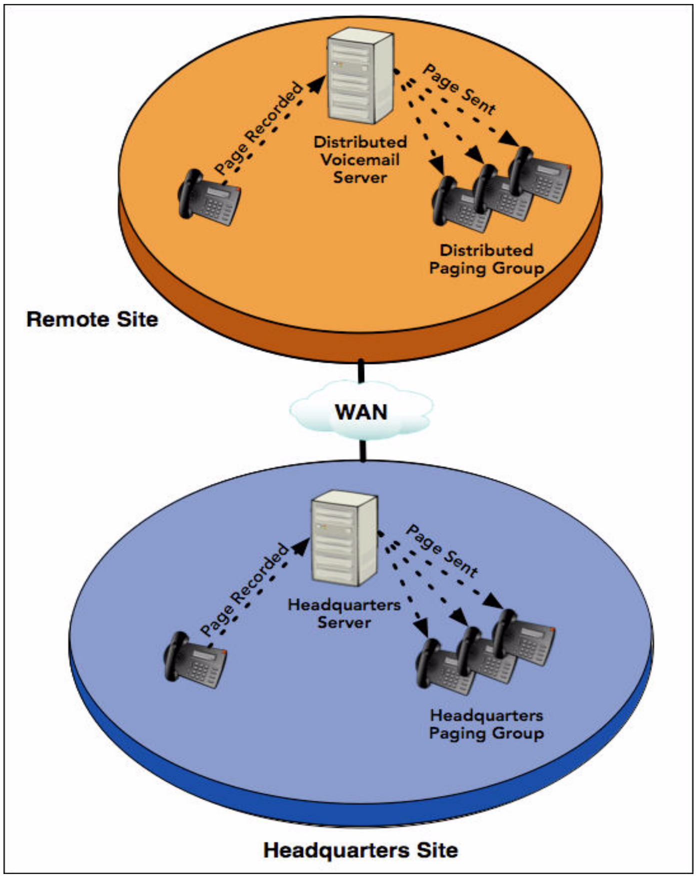


Figure 9-27 Multi-site Paging Groups

NOTE There is no additional licensing requirement to implement Multi-site Paging Groups. This feature is not implemented on ShoreTel Voicemail Switches.

9.9.1 Paging Groups Operation

To use Paging Groups, a person picks up a phone and dials the Paging Group extension. They then record their page message and hang up the phone. The Paging Group Sever then attempts to play the page on each phone referenced by the Extension List configured for the Paging Group.

In order to reduce the perceived audio delay of paging multiple phones in the same room, the ShoreTel system will wait to verify that the extensions referenced on the Extension List are ready to receive the page. This delay period is specified in the Group Paging Synchronization Delay field in the Paging Groups edit page.

9.9.2 Configuring Paging Groups

Once the desired Extension Lists are created, the administrator can then assign the Extension Lists to Paging Groups. The following steps outline the procedure to configure a Paging Group:

Step 1 Log in to ShoreTel Director.

Step 2 Select **Call Control** under the **Administration** section.

Step 3 Click the **Paging Groups** link to display the list of Paging Groups as shown in figure below.

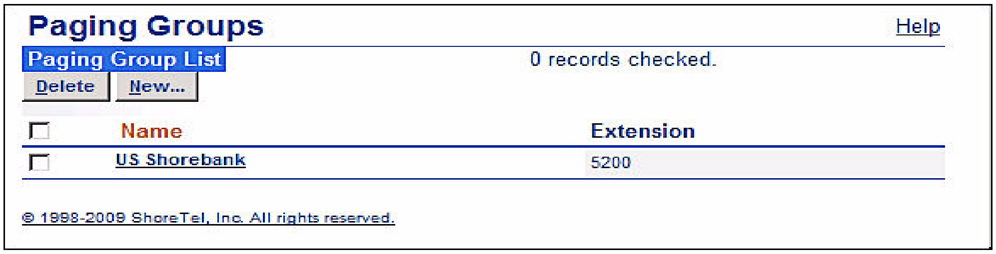


Figure 9-28 Paging Groups List Page

Step 4 Click **New** to create a new paging group or click on an existing Paging Group to modify the parameters.

Step 5 Edit the **Paging Groups** fields.

Step 6 Click the **Save** button to save your changes.

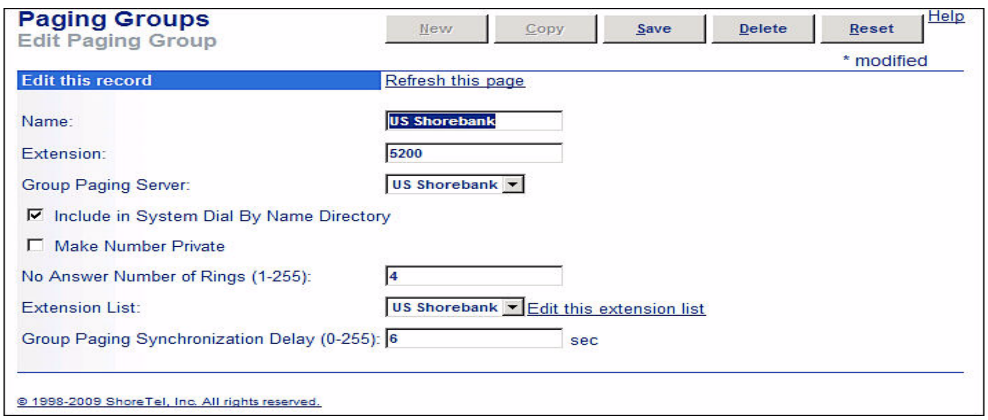


Figure 9-29 Paging Groups Edit Page

- **Name:** The paging group. Each paging group name must be unique. This is a required field.
- **Extension:** The extension number of the paging group. Each paging group extension must be unique. This is a required field.
- **Group Paging Server:** Select the Headquarters Server or one of the Distributed Voicemail Servers. The server selected will be the source of the audio streamed to the telephones during the page. The default server is set to Headquarters. This is a required field.
- **Include in System Dial By Name Directory**: This check box includes the paging group in the auto-attendant dial-by-name directory. No name is recorded for a paging group. When a paging group is chosen, the extension is announced by a generic message. **This is an** optional field.
- **Make Number Private**: This check box makes the paging group extension private. When the paging group is private, it is not listed in the system directory and does not appear in the Communicator dialing lists. T**his is a**n optional field.

- **No Answer Number of Rings**: The default is 2 rings. This is always used for analog phones. It is used for IP phones if the phone is busy. This is a required field.
- **Extension List**: Select the name of the extension list to be used as the paging group from the drop-down list. This is a required field.
- **Group Paging Synchronization Delay**: The time in seconds that the server will wait to connect to all phones in the extension list prior to sending the audio stream to the phones. This delay was introduced to reduce the perception of audio echo when paging large groups of phones. The default time is set to 3 seconds. This is a required field.

9.9.2.1 Important Considerations

- The maximum number of extensions that can be paged at one time is 100.
- Group paging is not available to external callers.
- Group paging is not available on Voice switches.

9.10 Priority Group Paging

Many organizations rely on paging for critical communications with their employees. In business, calls that are made to an IP phone from a paging server will show up as normal or a non-urgent call to the user when the phone is in use. In these instances the individual being paged has no idea of this request and there is no guarantee that the individual will suspend their current call to listen to the page being delivered.

Priority Group Paging provides system administrators with a method for designating groups of extensions that can be paged by dialing a single system extension and recording a message.

9.10.1 How Priority Group Paging Works

When Priority Group Paging is enabled, the recipient of a Priority Group Page or forced page will hear the audio of the page regardless of whether or not they are on a call. If the intended recipient is on an active call, that call will be automatically be placed on hold before the page is played. When the page completes, the call that was placed on hold will automatically resume.

Priority paging differs with Paging Groups in the way a Hold/Transfer/Conference is handled. A normal Paging Group page call will terminate when the user presses Hold/ Transfer/Conference. During a priority page this operation is not allowed and will be ignored.

9.10.2 Enabling Priority Paging Groups

Priority paging is an enhancement to the existing implementation of page groups by allowing the server to act as a media relay from source to the recipient. The call controller or the switch plays a limited role.

NOTE The Priority Paging feature provides new functionality to the page recipient. There are no changes in the core paging implementation.

To enable priority paging groups functionality.

Step 1 Log into ShoreTel Director.

Step 2 Click **Administration > Call Control > Paging Groups**. The Paging Groups window opens.

Step 3 Click the **New** button to create a new Paging Group. The Edit Paging Group page appears as shown in Figure 9-30.

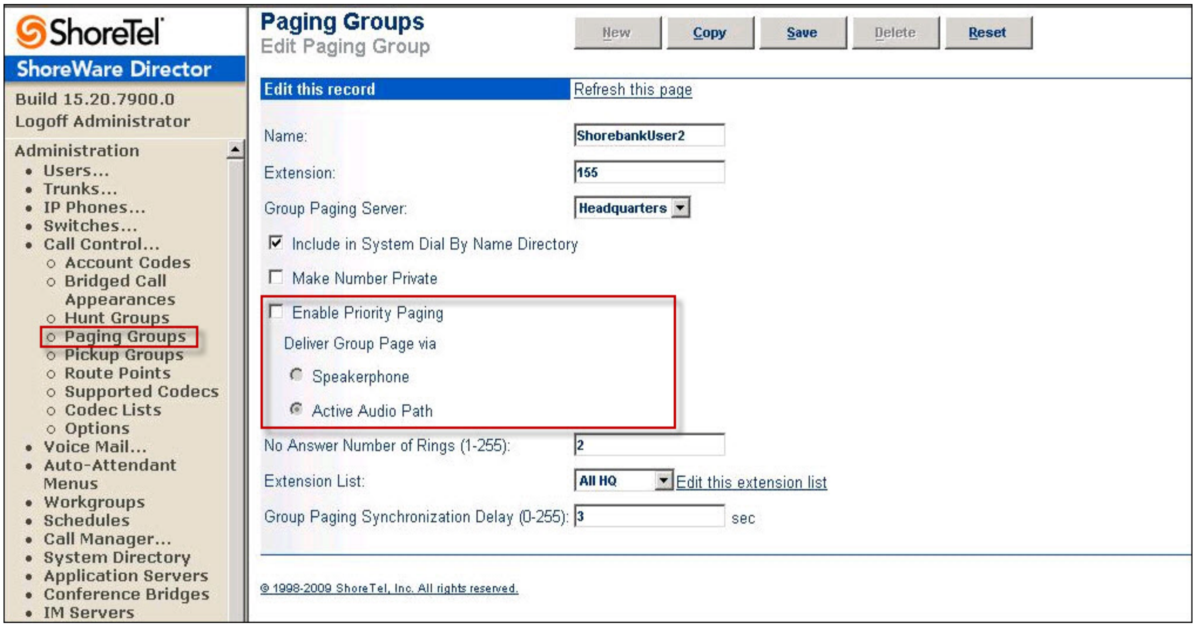


Figure 9-30 Paging Groups Edit Page

Step 4 Edit the following fields in the **Edit Paging Groups** screen.

- **Paging Group Server** - The server hosting the Workgroup. All Distributed Voicemail Servers will be listed. Voice Model Switches will not be listed.
- **Make Number Private** -This check box makes the paging group extension private. When the paging group is private, it is not listed in the system directory and does not appear in the Call Manager dialing lists.
- **Enable Priority Paging** -This checkbox enables Priority Group Paging. The default is unchecked.
- **Deliver Group Paging via** -This check box allows the administrator to configure preferred audio path on the phone to deliver the page.
    - **Speakphone** - The Page is played on the speaker.
- **Active Audio Path** - Active Audio path refers to the active media source such as a headset or phone handset.
- **Backup extension** -The backup extension of the Workgroup. If the Workgroup does not answer after the specified number of rings (for example, server unavailable or network problem), the call is routed to this extension. This allows you to configure backup call routing in case of failures.
- **No Answer Number of Rings** - This is field is required. Default is 2 rings. This is always used for analog phones. It is used for IP phones if the phone is busy.
- **Extension List** - Select the name of the extension list to be used as the paging group from the drop-down list.

- **Group Paging Synchronization Delay-** The time in seconds that the server will wait to connect to all phones in the extension list prior to sending the audio stream to the phones. This delay was introduced to reduce the perception of audio echo when paging large groups of phones. The default time is set to 3 seconds. **This is a required field**.

Step 5 When finished, click Save.

9.10.2.1 Important Considerations

- When a previously held call is restored, the audio path of the original call is retained. However, in a special case—when the page is over a speaker phone and the user decides to “cradle” the handset, for example—the audio path is not restored to the previous audio path. The audio path would be speaker phone. This exception is for Handset only—the headphone should work as expected. The problem with Handset is that, today’s ShoreTel IP Phones are not notified when the user puts the handset in the cradle (and is on speaker).
- Page-over-Page is not supported, i.e. if you issue a Priority page to an extension, while the extension is already being paged, the page is presented as an incoming call. This is because we do not have priorities set for page groups. In other words, if there were priorities assigned to PGs a page-over-page would result in a lower priority page being put on hold and the higher priority page being auto answered.
- Other Paging group limitations apply. SIP/Analog/OAE page will still be delivered but the calls cannot be auto answered.

9.11 Pickup Groups

Pickup Groups is a traditional PBX and key system feature used in group environments that allows users to answer any ringing phone in that group. The feature works best in places where a set of people work together on a daily basis, such as design firms. If a group member is away from her desk and across the room when her phone rings, she can quickly answer the call from another person's IP phone by pressing the relevant soft key or programmable button, or by using a simple star command (*13 + extension) from an analog phone.

Similarly, if she is out of the office and her phone rings, anyone can answer her call from another phone with a simple 'group pickup' command and take a note for her.

User extensions are added to an extension list and then this list is associated with a pickup group. The pickup group has its own extension (e.g. x3755), but this extension is invalid, cannot be dialed, and thus acts more like a code than an actual dialable extension.

- Pickup groups can be associated with a programmable toolbar button, or with a programmable button on an IP phone, or on IP phones that have soft keys.
- The user whose phone will be picked up must have class of service “Call Pickup Allowed” to use this feature. However, other users need not be members of the pickup group to pickup a call.
- This feature is not supported on the following legacy ShoreTel switch models: ShoreTel T1 and ShoreTel E1.
- The pickup feature will support:
  - 24 members per group
  - 16 groups per switch

— The members assigned to all pickup groups on a switch cannot exceed 80

— A single user can be a member of up to 5 pickup groups

- A single switch can host a combined total of up to 24 hunt groups, bridged call appearances, and pickup groups.
- This feature can be accessed in three different ways:

— **IP Phone –** If a programmable button has been configured for this feature, the user can press the button, or key, and enter the extension for the pickup group to answer the call.

— **Communicator –** If one of the pre-programmed buttons in Communicator has been set up for pickup groups, a user can enter the extension of the group to answer the call. If the key has already been programmed with the extension of the pickup group, then it is not necessary to enter the extension.

— **Analog Phone –** The user can enter the **13* command from the keypad, followed by the pickup group extension to answer calls from an analog phone.

9.11.1 Configuring Pickup Groups

Configuring the Pickup Groups feature consists of two separate tasks. First, you must create an extension list and populate it with the extensions of the members that will belong in this group. Second, you must create and name the pickup group and associate it with the extension list you just created.

To create the new extension list for a pickup group:

Step 1 Launch ShoreTel Director and enter the user ID and password. Then click the Login button.

Step 2 Click **Administration > Users > Extension Lists**. The Extension Lists page appears.

Step 3 Click the **New** button to to create a new extension lists. The Edit Extension List page appears as shown in Figure 9-31.

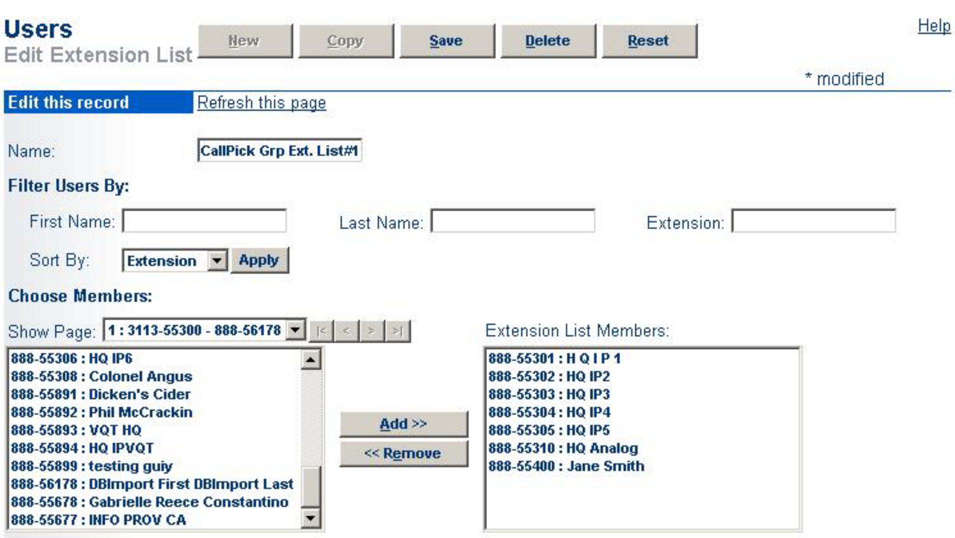


Figure 9-31 Edit Extension List Page

Step 4 Enter the desired name in the **Name** field.

Step 5 Select one or more users from the list on the left pane, and click the **Add** button to move them over to the **Extension List Members** pane on the right.

Step 6 Click **Save** to store your changes.

To create a new pickup group and associate it with an extension list:

Step 1 Launch ShoreTel Director still open, click on the **Call Control** link.

Step 2 Click on the **Pickup Groups** link to display a window similar to the one shown in Figure 9-32:

Figure 9-32 Pickup Groups Page

Step 3 Click the **New** button to display a window similar to the one shown in Figure 9-33:

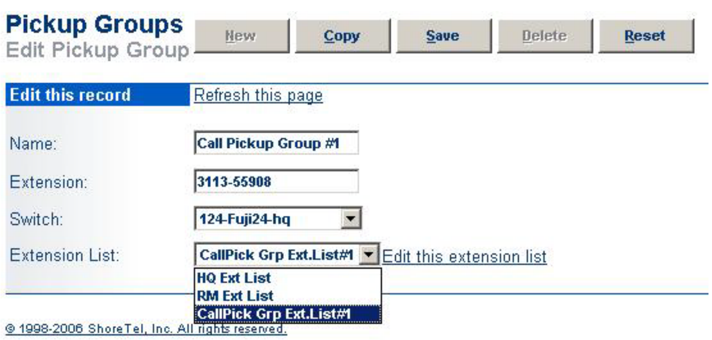


Figure 9-33 Edit Pickup Group

Step 4 Enter a name for the new Pickup Group in the **Name** field.

Step 5 The **Extension** field will auto-populate.

Step 6 Click on the **Switch** drop-down menu and select the appropriate switch for this group. You should select the switch that is physically closest to the members of the group.

Step 7 Click the **Extension List** drop-down menu and select the extension list that you just created in the previous task.

Step 8 Click **Save** to store your changes.

9.12 Route Points

Route points allow third-party applications complete access to call control signalling (using TAPI) and the actual voice media stream (using TAPI and WAV APIs). Configuring a route point enables calls to be terminated and controlled by a server on the network. To configure a route point, click the Route Points link under Call Control in the navigation frame. The Route Points list page is shown in Figure 9-34.

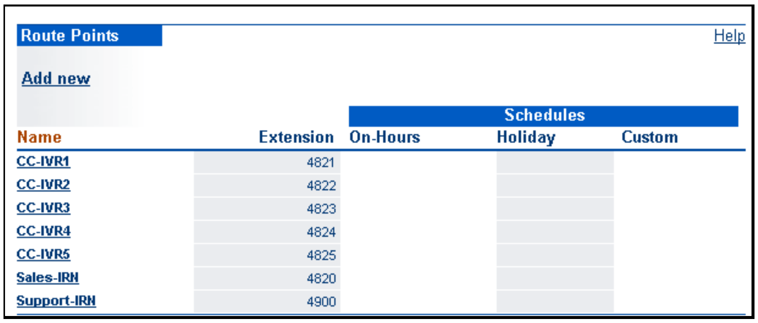


Figure 9-34 Route Points List Page

Then click "Add new" to invoke the Route Point edit page (shown in Figure 9-35).

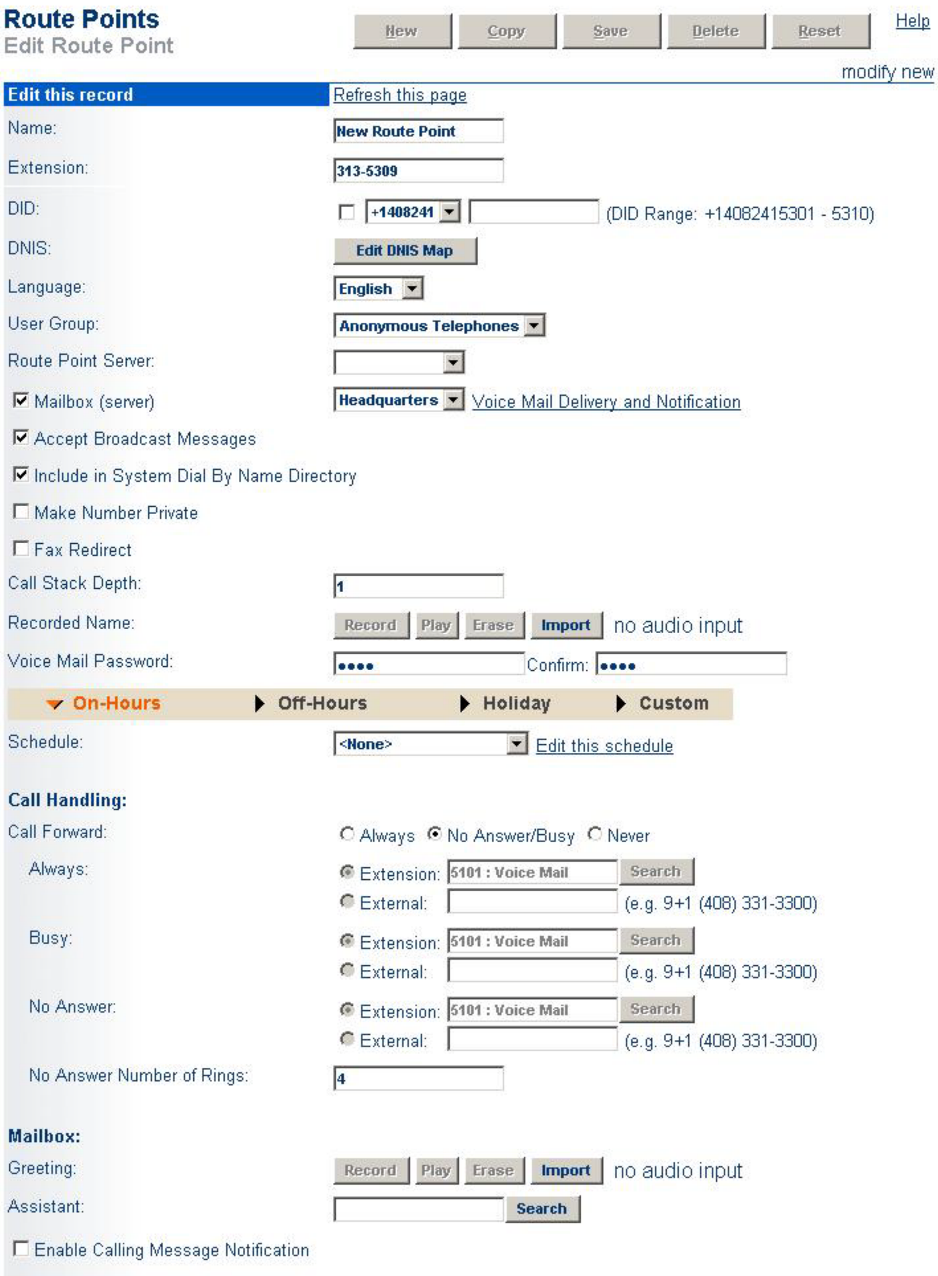


Figure 9-35 Route Point Edit Page

9.12.1 Parameters

The parameters that appear on the ***Route Point*** edit page are as follows:

- **Name:** This is the name of the route point.
- **Extension:** This is the extension number of the route point. Each route point extension must be unique. This is a required field.

  If you change the existing extension number to a new number and there is an associated mailbox, messages will be retained.
- **DID**: You can assign one DID number to a route point. This is an optional field.

Refer to "DID Ranges" on page -164 for additional information about DID numbers and ranges.

- **DNIS:** The Edit DNIS Map button invokes the Select DNIS Trunk Group dialog box. This lets you select a trunk group for DNIS routing. Only trunk groups that are configured for DNIS will be presented in the dialog box. You can assign multiple DNIS numbers to a workgroup. This is an optional field.

  DNIS is typically used to route 800-number calls to a workgroup or application.

- **Language:** Select the route point language from the drop-down list.
- **User Group:** This drop-down list lets you assign a user group to the route point.

  The route point requires permissions just like a user. For instance, the route point call to external call forwarding needs access to trunk groups and has a mailbox. This is a required field.

  Consult Section 10.3.

- **Route Point Server:** This selects the server that provides route point services for third-party applications. Third-party applications gain control of calls handled by the ShoreTel system through route points. This is a required field.

  It is recommended that the route point server be a separate server from the headquarters server and not be configured with mailboxes.

- **Mailbox (server):** This provides the route point with a mailbox on the associated server. If you change the server, all messages are automatically moved to the new server. The default mailbox server is the headquarters server. This is a required field.
- **Accept Broadcast Messages:** This check box enables the route point mailbox to receive broadcast messages. This is an optional field.
- **Include in System Dial by Name Directory:** This check box includes the route point in the auto-attendant dial-by-name directory. This is an optional field.
- **Make Number Private:** This check box makes the route point extension private. When the route point is private, it is not listed in the system directory and does not appear in the Communicator dialing lists. This is an optional field.
- **Fax Redirect:** This check box enables fax redirection. When the route point answers a call and a fax tone is detected, the fax is redirected away from the route point to the headquarters fax extension. This is an optional field.
- **Call Stack Depth:** This lets you specify the maximum number of simultaneous calls that can be "stacked" on the route point extension. When this number is met, additional inbound calls will be routed to the call forward busy destination.

  Valid entries are 1 through 200.

  Call Stack depth is a licensed feature.

- **Recorded Name:** The Record, Play, Enter, and Import buttons let you record a name for the route point. The Recorded Name is used as part of the default mailbox greeting as well as in the dial-by-name directory. This is an optional field.

  You can use your PC microphone and speakers or a telephone to play and record within ShoreTel Director. Please refer to the auto-attendant options for more information.

  You can also import prompts into ShoreTel Director. Prompts must be recorded as µ-law WAV files.

- **Voice Mail Password:** This is the password used for accessing a route point voice mailbox over the telephone. Enter a password in the first text-entry field and again in the Confirm field. This is a required field.

  The default password is “1234”. Passwords may be numeric only.

- **Schedule:** You can configure schedules for the On-Hours, Holiday, and Custom modes that automatically change the call handling of the route point. The rules for schedules are:

  — If it is custom time, use Custom mode.

  — If it is holiday time, use Holiday mode.

  — If it is on-hours time, use On-Hours mode.

  — Otherwise, use Off-Hours mode.

  If no schedules are specified, On-Hours mode is used.

  The Edit this schedule link provides a quick way to navigate to the associated schedule. This is an optional field.

- **Call Handling Call Forward:** These buttons let you specify when calls are forwarded. The conditions are Always, No Answer/Busy, and Never.

  — **Always—**The Always condition forwards calls to the number specified in the Always Destination parameter immediately when a call is received.

  — **Busy—**The Busy condition forwards calls to the Busy Destination immediately if the user’s call stack is full.

  — **No Answer—**This condition forwards calls if there is no answer.

  — **No Answer Number of Rings—**Sets the number of rings after which a no answer condition is assumed to exist.

  This is a required field.

  — **Always Destination—**When the Always call forward condition is selected, calls are forwarded immediately to this extension. You can also forward calls to an external number (access code required).

  — **Busy Destination—**When the No Answer/Busy call forward condition is selected, calls are forwarded to this extension immediately if the user’s call stack is full. You can also forward calls to an external number (access code required).

  — **No Answer Destination—**When the No Answer/Busy call forward condition is selected, calls are forwarded to this extension after the specified number of rings. You can also forward calls to an external number (access code required).

- **Mailbox Greeting:** This lets the user record a greeting for his or her mailbox, using the Record, Play, Erase, and Import buttons. This is on by default.

- **Assistant:** Each route point can have a personal assistant, which is the destination a calling party is transferred upon dialing “0” in the route point mailbox. This is an optional field.

- **Enable Calling Message Notification:** This check box enables message notification for this call handling mode. The manner in which the user is notified is determined by the user’s message notification settings. The recommended default is off.

9.13 Call Control Options

Set the general call control options from the Call Control Options page, as shown in Figure 9-36.

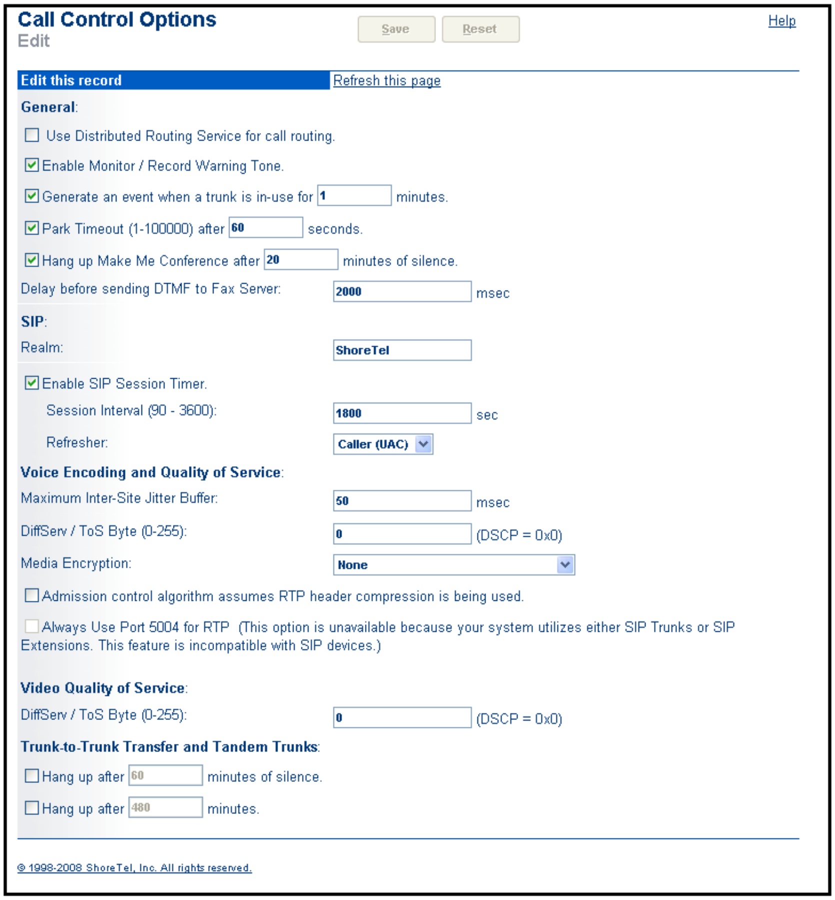


Figure 9-36 Call Control Options Edit Page

9.13.1 Distributed Routing Service

Distributed Routing Service (DRS) lets a large system scale beyond 100 switches to up to a total of 500 switches (including SoftSwitches). The DRS is optional on systems up to 100 switches but must be enabled on systems with 101 or more switches.

When Distributed Routing Service is disabled, ShoreTel switches in a system build an internal routing database from the peer-to-peer communication with other switches. Each ShoreTel switch contains routing information for all endpoints in the system including information regarding trunk selection for outbound calls. When calls are placed from any extension, each switch is able to route the call to the correct ShoreTel switch based on its internal routing database.

When Distributed Routing Service is enabled, ShoreTel switches only exchange routing information with other switches at the same site rather than exchanging routing information with every other switch in a multi-site system. Although each ShoreTel switch only maintains routing information within its site, each ShoreTel server also includes an instance of the Distributed Routing Service which maintains system-wide routing information. When calls are initiated, ShoreTel switches contact the Distributed Routing Service in order to find the ShoreTel switch or switches needed to complete the call.

In a system with more than one ShoreTel server, the ShoreTel switches may contact an alternate instance of the routing service if the primary instance is not reachable. ShoreTel servers have a hierarchical relationship and switches first try to contact the nearest instance of the Distributed Routing Service in the hierarchy. If that instance of DRS is not reachable, the instance of DRS at the parent server in the hierarchy will be contacted as a fallback. If both instances of DRS are not reachable the switch will make a best effort to route the call based on its internal routing tables built from communicating with peer ShoreTel switches at the same site.

9.13.2 Parameters

The parameters on the Call Control Options edit page are as follows:

9.13.2.1 General Parameters Area in the Options Window

- **Use Distributed Routing Service for call routing:** Enables Distributed Routing Service (see preceding explanation.)
- **Enable Monitor / Record Warning Tone:** When a 2-way or Make Me Conference call is monitored or recorded, checking this box causes a tone to be played that warns of monitoring or recording. This option is enabled by default.

  Deselect this check box to enable silent recording. Silent recording allows operators and supervisors to hide the fact that they are recording agents' calls by “silently” recording those calls. This behavior can be desirable in certain situations, such as for monitoring the telephone manners of an employee.

  When the recording is silent or hidden, Communicator offers no visual or audible indication that the call is being recorded. The periodic beeping sound (used to notify call participants that their calls are being recorded) is suppressed.

Example: ShoreTel does not warrant or represent that your use of call monitoring or recording features of the Software will be in compliance with local, state, federal or international laws that you may be subject to. ShoreTel, Inc. is not responsible for ensuring your compliance with all applicable laws.

Before disabling the warning tone, you may wish to consult with legal counsel regarding your intended use.

- **Generate an event when a trunk is in use for N minutes:** You can set this parameter to generate an event log when a trunk has been in use for the specified time period.
- **Park Timeout after NNNNNN seconds:** You can set how long a call will remain on park before the call returns to the party that parked the call. The timeout is in seconds and can have a value from 1 to 100,000 seconds. Unchecking the Park Timeout check box allows calls to be parked indefinitely.
- **Hang Up Make Me Conference after NN minutes of silence:** The default timeout is 20 minutes. If a conference is silent for the set length of time, a hang-up will be forced.
- **Delay before sending DTMF to Fax Server:** Enter the amount of delay (in milliseconds) after which the DTMF information will be sent to the fax server. Consult your fax server documentation to find out how much delay the fax server expects.

9.13.2.2 SIP Parameters in the Call Control Options Window

- **Enable SIP Session Timer:** Select this check box to have keepalive heartbeats sent to SIP endpoints.
- **Session Interval (90-3600):** Enter the number of seconds in the Session Interval field to specify the keepalive interval at which heartbeats will be broadcast. The heartbeat will be sent out at the specified period and if no response is received, the session will be dropped. (See RFC 4028 for details on this parameter.)
- **Refresher:** Use the Refresher drop-down menu to specify whether the SIP Session Timer will be applied to the caller or callee. Choices are None, Caller UAC, and Caller UAS. (See RFC 4028 for details on this parameter.)

9.13.2.3 Voice Encoding and Quality of Service in the Call Control Options Window

- **Maximum Inter-Site Jitter Buffer:** This parameter sets the maximum size of the jitter buffer. A larger jitter buffer might result in more delay between calling parties, which might degrade the quality of service. The buffer can be set from 20 to 300 msec. The default is 50 msec.
- **DiffServ/TOS Byte:** This parameter configures DiffServ/ToS for voicemail, workgroup, account code collection (ACC), and contact center calls.The value entered in this field must be a decimal number between 0 to 255. This setting applies to all ShoreTel servers in a ShoreTel system. To enable a new DiffServ/ToS setting, you must reboot all ShoreTel servers.

  NOTE ToS is an eight-bit field in the IP Packet Header. It is used to accommodate applications that require real-time data streaming, as specified by RFC 3168. The ToS fields contains a six-bit Differentiated Services Code Point and a two-bit Explicit Congestion Notification field.

- **Media Encryption:** This option specifies the encryption method used by ShoreTel to protect payload packets. Refer to the "Media Encryption" on page 310 for more information.
- **Admission control algorithm assumes RTP header compression is being used:** To enable this feature, select the check box.
- **Always Use Port 5004 for RTP:** By default, this box is checked. This option is not available on systems that utilize SIP extensions or SIP trunks.

  All IP Phones and switches must be rebooted for settings to take effect. Failure to reboot may result in one way media.

9.13.2.4 Video Quality of Service in the Call Control Options Window

- **DiffServ/ToS Byte:** This parameter configures the DiffServ/ToS field in the IP Packet Header of the Video Call payload packet. The default value for the Video DiffServ/ToS Byte is 0, indicating that priority is not defined for video. Changing this setting does not affect active video sessions. The updated value is applied to new video sessions. Communicator recognizes the new value without being restarted.

9.13.2.5 Trunk-to-Trunk Transfer and Tandem Trunks in the Call Control Options Window

This lets you manage trunk-to-trunk transfers when the Allow Trunk-to-Trunk Transfer parameter is enabled on the Telephone Features Permissions class of service edit page. The field is defined in the section called Section 10.2.1.

The ShoreTel system supports trunk-to-trunk transfers, which allow a user to transfer an external caller to an external number. Since this feature can lead to unwanted toll charges, the system also supports a class of service permission that only grants this feature to selected user groups. To grant this permission, you enable the Allow Trunk-to-Trunk Transfer parameter on the Telephone Features Permissions class of service edit page.

Users with trunk-to-trunk transfer permission might accidently transfer an external caller to an external number without realizing it. This can lead to “hung” trunks, resulting in the inability to make outbound calls or take inbound calls.

The ShoreTel system lets you manage trunk-to-trunk transfers using the Call Control Options page. This lets you eliminate unwanted trunk-to-trunk transfers while ensuring that valid trunk-to-trunk transfers are not dropped.

The following are considered trunk-to-trunk transfers:

- A user is talking with an external party and transfers the external party blindly or consultatively to an external number.
- A three-party conference call is taking place with one user and two external parties, and the user drops from the call.

An external party forwarded to an external number by a user’s call handling mode is not a trunk-to-trunk transfer.

- **Hang up after N minute(s) of silence:** Enabling this parameter automatically drops trunk-to-trunk transfers after both parties have been silent for the defined time period. The default time period is 60 minutes.
- **Hang up after N minutes:** Enabling this parameter automatically hangs up the trunk-to-trunk transfer after the defined time period. This parameter should be set only if truly needed and set for a long period. The default period is 480 minutes.

9.14 Bandwidth Management and Codec Negotiation

ShoreTel supports additional codecs available on ShoreTel IP Phone and through SIP devices. Codec negotiation during voice call setup is facilitated by data structures, including codec lists and profiles.

Bandwidth Management and Codec Negotiation tools available through Director include:

- Codec lists that are configurable through Director.
- Video codec support
- A Director parameter that permits inter-site video sessions.
- SIP codec negotiation method that complies with RFC 3264 – An Offer / Answer Model with the SDP.

9.14.1 Codec Lists

Codec lists enumerate a set of codecs. Director defines two types of codec lists: Supported Codecs and Preferred Codecs.

9.14.1.1 Supported Codecs

Supported Codecs is a comprehensive list of all codecs available to system devices. When ShoreTel is initially installed, the Supported Codecs list comprises the set of codecs provided by ShoreTel and available on ShoreTel IP Phones. Administrators can add codecs to support SIP devices that may use codecs not initially provided by ShoreTel.

The Supported Codecs list also indicates the bandwidth required by each codec. The bandwidth numbers are used by ShoreTel to allocate bandwidth as voice calls are initiated and terminated.

Supported Codecs list contents, including the bandwidth settings, are passed to all switches in the system, where they are used for selecting codecs for individual call sessions.

9.14.1.2 Preferred Codec Lists

Preferred Codec lists are subsets of the Supported Codecs list. Preferred Codec lists are referenced by Sites within a ShoreTel system to specify the codecs used for intersite and intrasite voice calls. ShoreTel provides six default codec lists which cannot be deleted or modified. Administrators can define additional lists through Director.

The default codec lists provided with ShoreTel include:

- Fax Codecs
- High Bandwidth Codecs
- Low Bandwidth Codecs
- Medium Bandwidth Codecs
- Very High Bandwidth Codecs
- Very Low Bandwidth Codecs

9.14.2 Intersite Video Sessions

Intersite Video Sessions are enabled through a Telephony Class of Service setting. ShoreTel does not allocate bandwidth for video calls.

9.14.3 Codec Negotiation

ShoreTel supports simultaneous audio, video, and data codec negotiations to facilitate multimedia sessions between SIP endpoints as defined by RFC 3264 – An Offer / Answer Model with the SDP. Codecs specified in the Supported Codec list are offered during session parameter negotiations.

The ShoreTel negotiation process that supports RFC 3264 is as follows:

1. The calling device sends the list of codecs it supports to the ShoreTel servicing the call.
2. The ShoreTel switch that controls the calling device compiles a codec list. The codec list contains all codecs contained in the following:
   — The calling device's codec list
   — One of the site's codec lists – intersite, intrasite, or fax – depending on the call type

Codecs on the combined list are sorted as specified by the selected site codec list.

3. (Intersite Calls only): The ShoreTel switch that controls the destination device modifies the codec list by removing all codecs that are not listed on the destination site's codec list.
4. The ShoreTel switch controlling the destination device sends the codec list to the destination device.
5. The destination device replies by sending a list of one or more codecs to the originating device.
6. The list sent by the destination device typically includes, from the received codec list, the highest priority codec it can support.
7. The two devices begin sending RTP streams using the highest priority codec listed in the destination device's reply.

9.14.4 Implementation

9.14.4.1 Supported Codecs Page

The Supported Codecs page lists the audio codecs that are available to devices making voice calls through ShoreTel. Codec lists used when negotiating call parameters comprise codecs displayed on this page. Although most commonly used codecs are listed on this page, the system can also support other codecs by adding them to the list.

To access the Supported Codecs page, shown in Figure 9-37, select **Administration > Call Control > Supported Codecs** from the Director menu. Each row in the table lists a codec.

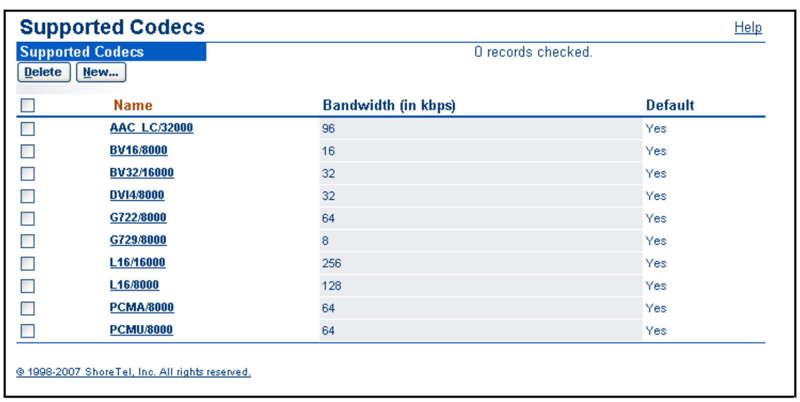


Figure 9-37 Supported Codecs Page

Table fields in the Supported Codecs page include:

- **Name:** This parameter is fully qualified codec ID string of the codec. ShoreTel uses this string to specify codecs while negotiating with other calling devices.

  The codec ID string comprises the name and sampling rate of the codec. While the codec name generally corresponds with the name by which the codec is commonly referred, PCMA specifies a G.711 codec (A-law) and PCMU specifies a G.711 codec (μ-law).

- **Bandwidth (in kbps):** This parameter identifies the bandwidth required by the codec. ShoreTel uses this figure when allocating bandwidth resources.
- **Default:** This parameter specifies the source of the codec entry:

— **Yes** indicates a codec that was provided with ShoreTel. Default codecs cannot be removed from the list.

— **No** indicates a codec that was provided by the system administrator. These codecs can be removed from the list.

To add a new codec to the list, do the following:

Step 1 Click the Add button at the top of the page to open the Supported Codec Info popup, shown in Figure 9-38.

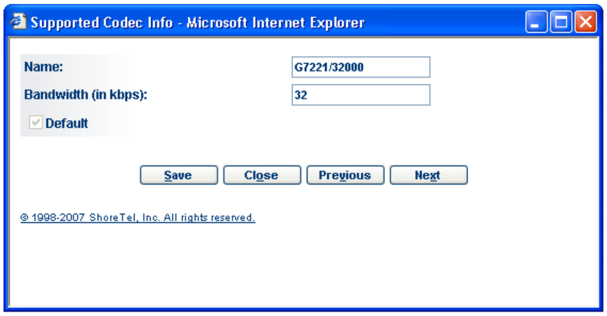


Figure 9-38 Supported Codec Info Popup

Step 2 Enter the fully qualified codec ID string in the Name data entry field.

The field must be entered exactly as expected by devices that negotiate call parameters.

Step 3 Enter the codec bandwidth in the Bandwidth data entry field.

Entering incorrect numbers in this field compromises ShoreTel's ability to manage bandwidth resources.

Step 4 Press the Save button to store the changes to the database and return to the Supported Codecs page.

The Default field is used when editing existing codecs to denote a codec that is supplied with ShoreTel. This parameter is not editable.

To edit an existing codec:

Step 1 Click the name of the codec to be edited to open the Supported Codec Info popup.

Step 2 Change the contents of the Name or Bandwidth data entry fields.

The Default box is not editable.

Step 3 Press SAVE to store the changes and return to the Supported Codecs page.

9.14.4.2 Codec Lists

Codec lists are subsets of the codecs supported by the ShoreTel system. Codec lists are selected in the Edit Sites page to determine the codecs used for intersite and intrasite calls. Director pages that support codec lists include the Codec List page and the Edit Codec Lists page.

Each codec list must include the PCMU (G.711 µ-law) codec.

Codec List Page

The Codec List page, shown in Figure 9-39, provides a roster of Codec Lists configured on the system. Codec list names, as specified on this page, are used on the Edit Sites page when selecting codecs for intersite and intrasite calls. To access the Codec List page, select Administration > Call Control > Codec Lists from the Director menu. Each line in the table corresponds to a Codec List.

Figure 9-39 Codec List Page

To add a new codec list, open the Edit Codecs Lists page by clicking the New button in the top left corner of the page.

To view or edit an existing codec list, open the Edit Codec List page by clicking the name of the desired codec list. Default codec lists supplied with ShoreTel cannot be edited.

To delete an existing codec list, select the box left of the desired codec and press the Delete button in the top left corner of the page. Default codec lists supplied with ShoreTel cannot be deleted.

Edit Codec List

The Edit Codec List page, shown in Figure 9-40, modifies the Codec List specified by the Name field contents. The page displays two tables:

- The **Choose Codec** table lists the codecs available on the system that are not in the specified codec list.
- The **Codec List Members** table displays the codecs that are in the specified codec list.

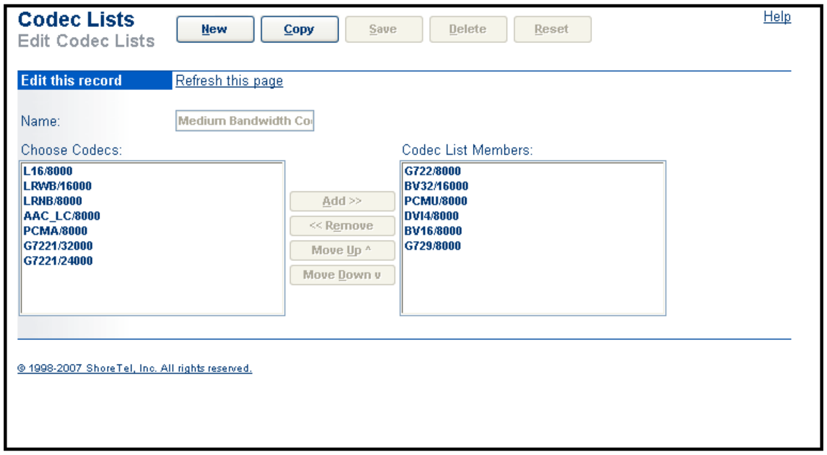


Figure 9-40 Edit Codec List Page

All codecs available on the system, as defined by the Supported Codecs page, are listed either in the Choose Codec table or the Codec List Members table.

To add a codec to the codec **list**, select the desired codec in the Choose Codec, then click the **ADD** button between the tables.

To remove a codec from the codec **list**, select the desired codec in the Codec List Members table, then click the **REMOVE** button between the tables.

To change the position of a codec within the Codec List Members table, select the desired codec in that table and click the **MOVE UP** or **MOVE DOWN** buttons located between the tables.

To save Codec List changes, click the SAVE button at the top of the page

To create a new codec list, press the **NEW** button at the top of the page.

To use the displayed codec list as a template for a new list, click the COPY button at the top of the page.

To revert the codec list to the last saved version, click the Reset button at the top of the page.

Default Codec Lists

ShoreTel provides six default codec lists that cannot be modified or deleted. The following enumerates the codecs, in order of priority, within each list:

- **Very High Bandwidth Codecs**
  - — L16/16000
  - — G722/8000
  - — BV32/16000
  - — L16/8000
  - — PCMU/8000
  - — DVI4/8000
  - — BV16/8000
  - — G729/8000

- **High Bandwidth Codecs**
  - G722/8000
  - BV32/16000
  - L16/8000
  - PCMU/8000
  - DVI4/8000
  - BV16/8000
  - G729/8000
- **Medium Bandwidth Codecs**
  - G722/8000
  - BV32/16000
  - PCMU/8000
  - DVI4/8000
  - BV/8000
  - G729/8000
- **Low Bandwidth Codecs**
  - BV32/16000
  - DVI4/8000
  - BV16/8000
  - G729/8000
  - PCMU/8000
- **Very Low Bandwidth Codecs**
  - G729/8000
  - PCMU/8000
- **Fax Codecs**
  - L16/8000
  - PCMU/8000

9.14.4.3 Intersite Video

To configure the Intersite Video setting for a Class of Service:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Users > Class of Service**.

Step 3 Click the name of the desired Telephony Features Permissions Class of Service in the top table of the page. The edit feature page appears as shown in Figure 9-41.

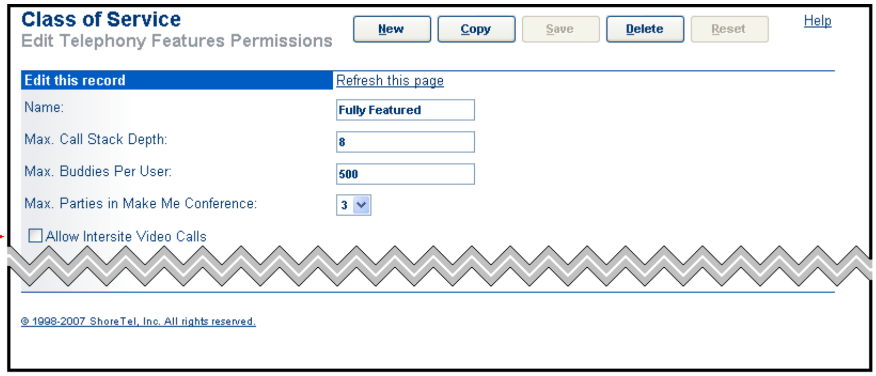


Figure 9-41 Allow Intersite Video Calls parameter

Step 4 Check the **Allow Intersite Video Calls** check box to allow members of the Class of Service to participate in video calls. Clear this setting to deny access to video calls.

9.15 Automatic Ringdown

An **automatic ringdown circuit** comprises predefined devices at the circuit endpoints and is configured to ring a recipient device immediately after an initiating device goes off hook. Automatic ringdown calls are completed without dialing or any other signaling other than the initiating device going off hook.

The simplest example of an automatic ringdown circuit consists of two points, with one telephone at each end of the circuit. When the telephone at one end goes off-hook, the phone at the other end immediately rings. Phones without dials may be used on each end of the circuit.

Ringdown circuits may be unidirectional or bidirectional.

- In unidirectional circuits, calls always originate from the same end of the network. A device on a receiving endpoint going off hook has no effect on the initiating endpoint device.
- In bidirectional circuits, both ends can initiate and receive automatic ringdown calls.

9.15.1 Dedicated Circuit Ringdown

ShoreTel IP Phones with call appearance buttons and the ShoreTel Button Box support Dedicate Circuit Ringdown. A call appearance button is programmed for Ringdown by configuring it as a Bridged Call Appearance and selecting a Ringdown option. When using ShoreTel IP Phones as the calling and recipient devices, an IP Phone button on both devices is be configured for Ringdown. ShoreTel supports using external devices as the ringdown recipient.

A ringdown call is initiated by pressing the ringdown button on a calling device. The ringdown button on the recipient device blinks; the recipient device can be configured to ring or remain silent on an inbound ringdown call. The ringdown call is answered on the recipient phone either by lifting the handset or pressing the blinking ringdown button. If the phone is configured to remain silent when a ringdown call is incoming, the blinking button must be pressed to answer the call.

ShoreTel IP Phones can be configured for one-way or two-way ringdown. When phones are configured for one-way ringdown, one phone is defined as the recipient device; pressing the ringdown button on that device will not initiate a call to the phone on the other end of the circuit. When phones are configured for two-way operation, pressing the ringdown button on either device initiates a call to the device on the other end of the circuit.

To force a device to ring until it is answered, the call stack depth for Bridge Call Appearances supporting ringdown should be set to 1 and call handling mode transfers should be disabled by selecting **No Answer.**

The following sections describe four typical ringdown configurations.

9.15.1.1 Basic One-to-One Ringdown

Basic one-to-one ringdown, as depicted in Figure 9-42, is enabled by configuring a Call Appearance button on each phone as a Bridged Call Appearance (BCA). During an incoming ringdown call, the BCA button on the recipient device blinks.

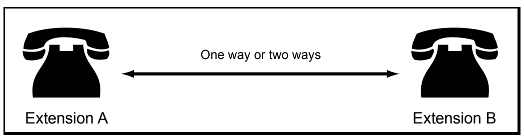


Figure 9-42 One to One Ringdown Operation

ShoreTel supports one-way or two-way ringdown in this configuration.

9.15.1.2 One-to-many Ringdown

ShoreTel supports one-to-many ringdown operation. When the Ringdown BCA is pressed on the calling device, Extension A in Figure 9-43, the corresponding button on all recipient devices programmed to receive the ringdown call flashes Green and off. When the call on one of the recipient devices is answered, the Ringdown button on the other devices is Red. If the answering device places the call on hold, the call is parked to the Bridged Call Appearance and is available for all of the recipient devices. Conference calls involving a ringdown number that is placed on hold is not parked onto the BCA extension.

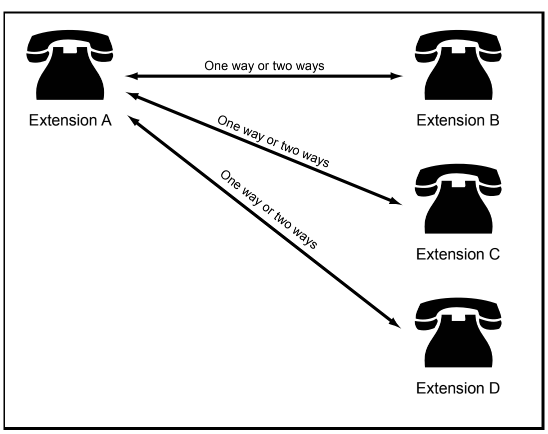


Figure 9-43 One to Many Ringdown Operation

ShoreTel supports two-way ringdown for the one-to-many configuration. When a ringdown call is placed from a recipient device, (Extension B in Figure 9-43), the BCA button on the other recipient devices (Extensions C and D) becomes solid red when Extension A answers the call, indicating that the line is busy. Pressing theses red buttons has no effect. If Extension A does not answer the ringdown call, the phone rings until the caller on Extension B hangs up or the call handling mode for Extension A handles the call.

9.15.1.3 Multiple ring down buttons

ShoreTel IP Phones with sufficient call appearance buttons support multiple ringdown circuits. In Figure 9-44, Extension A is configured to support two ringdown circuits:

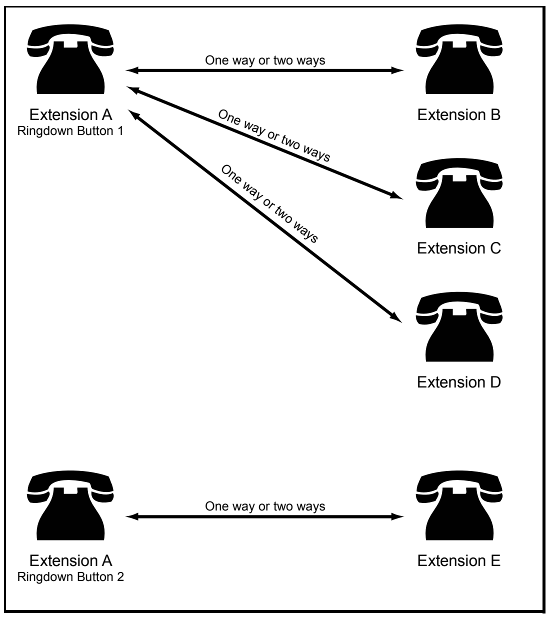


Figure 9-44 Multiple Ringdown Buttons

- One ringdown circuit with Extensions B, C, and D at the other end.
- One ringdown circuit with Extension E at the other end.

This configuration requires four Bridge Call Appearance extensions: two BCA extensions for Extension A; one BCA extension shared by Extensions B, C, and D; and one BCA extension for Extension E.

9.15.1.4 Ring Down to an External Device

Ringdown to an external device is supported by programming the ringdown button to access a trunk group that has a unique trunk access code and contains only one trunk. When the user presses the ringdown button, it accesses that trunk.

To place a call over a ringdown circuit to an external device, the user enters the trunk access code in addition to the phone number of the device. For instance, when configure a analog trunk group to service the ringdown call, the default trunk access code of 9 must be dialed to place a ringdown call.

This trunk is not required to be reserved solely as the ringdown circuit. Any user can dial this trunk access code to select this trunk. If the trunk is busy, pressing the ringdown button generates a busy signal. “Enable CHM” only applies for an incoming call. It is not applicable for an outbound call. To enable ringdown buttons on the ShoreTel devices (Extensions B, C and D in Figure 9-45), the BCA extension must be configured on the Trunk Groups page.

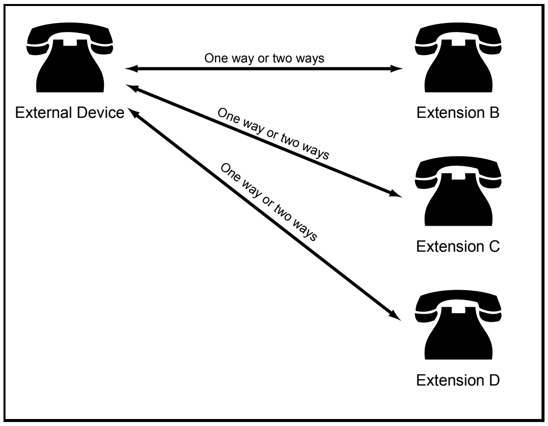


Figure 9-45 Ringdown from an External Device

9.15.2 Phone Delayed Ringdown

ShoreTel permits IP phones that provide a dial tone to perform ringdowns by taking the handset off hook. If a number is not dialed within a specified period after taking the phone off hook, a ringdown call is directed to the predefined recipient device.

When the ringdown device goes off hook by lifting up handset or pressing speaker or headset, the ringdown number is dialed if a digit is not entered within the configured ringdown delay period. If a digit is entered within this period, the ringdown call is not performed.

Analog phones to which a user extension is assigned can receive incoming calls even when configured for delayed ringdown calls. When an anonymous user is assigned to the port, the device cannot receive incoming calls if it is configured to make delayed ringdown calls.

9.15.3 Ringdown Implementation

Automatic ringdown circuits may also be configured in a one-to-many topology. In these circuits, one initiating device rings a group of phones on the other circuit endpoints.

Automatic ringdowns are activated on ShoreTel IP Phones by pushing an IP Phone Call Appearance button or lifting a handset on a dedicated ringdown device. These actions cause the recipient device to continuously ring until the call is answered or the calling party ends the call. The Call Handling Mode on the recipient device is ignored for ringdown calls. Automatic ringdown is supported only by phone; Communicator does not support automatic ringdown.

ShoreTel implements Ringdown through the following methods:

- **Dedicated Circuit Ringdown:** Ringdown is immediately initiated when a programmed Call Appearance button is pushed or a specified device goes off hook. The ringdown call is answered on the recipient by pressing the corresponding Call Appearance button or taking the device off hook.
- **Phone Delayed Ringdown:** ShoreTel permits IP phones that provide a dial tone to perform ringdowns by taking the handset off hook. If a number is not dialed within a

specified period after taking the phone off hook, a ringdown call is directed to the recipient device.

9.15.3.1 Director Configuration Pages

Ringdown circuits are implemented by configuring IP Phone buttons for circuit endpoints through Director. Delayed ringdown circuits are configured from the Edit User – General page.

Configuring IP Phone Buttons

IP Phone Buttons for ShoreTel users are configured from the Program IP Phone Buttons page, as shown in Figure 9-46. To access the Program IP Phone Buttons page, open the Edit User – Personal Options page for the desired user, then click Program IP Phone Buttons.

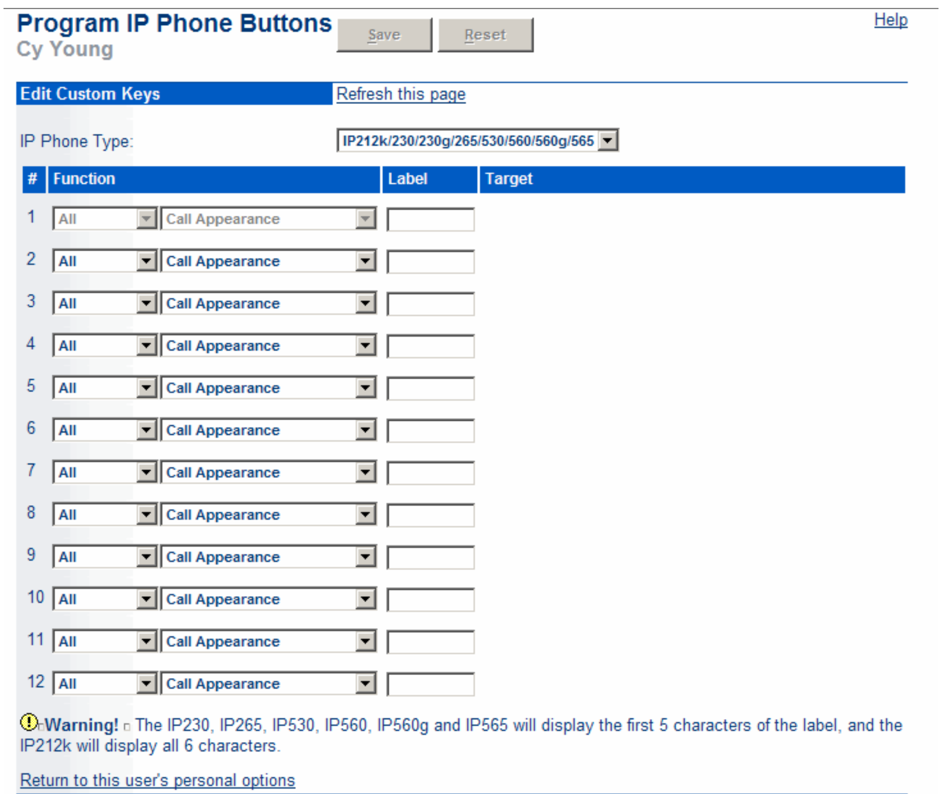


Figure 9-46 IP Phone Button Ringdown Options

In the figure above, button #2 is configured to monitor the Bridged Call Appearance specified by the Extension data entry field. Selecting **Answer Only** disables the ringdown feature on the specified button. BCA options located below the Normal Operation option configure ringdown options for the button.

BCA Ringdown parameters include:

- **Dial Tone:** Select this option to configure the phone as the recipient on a ringdown circuit. Buttons configured with Dial Tone cannot initiate ringdown calls.
- **Dial Extension:** Select this option to configure the button as the initiating end of a ringdown circuit when the recipient is an IP phone located on the ShoreTel network. The Extension data entry field specifies the Bridged Call Appearance that is dialed

when the IP Phone button is pushed. Buttons configured to perform ringdown calls can also receive ringdown calls.

- **Dial External:** Select this option to configure the button as the calling end of a ringdown circuit when the recipient is a device not located on the ShoreTel network. The data entry field specifies the phone number that is dialed when the IP phone button is pushed; the trunk access code required to access the trunk group dedicated to this ringdown circuit must be included as part of the phone number.

Configuring a Delayed Ringdown Circuit

Delayed ringdown circuits are configured on the Edit User – General page for the user extension that initiates the ringdown call. Figure 9-47 displays the section of the Edit – User General page containing the delayed ringdown parameter fields, which are located in the bottom third of the page.

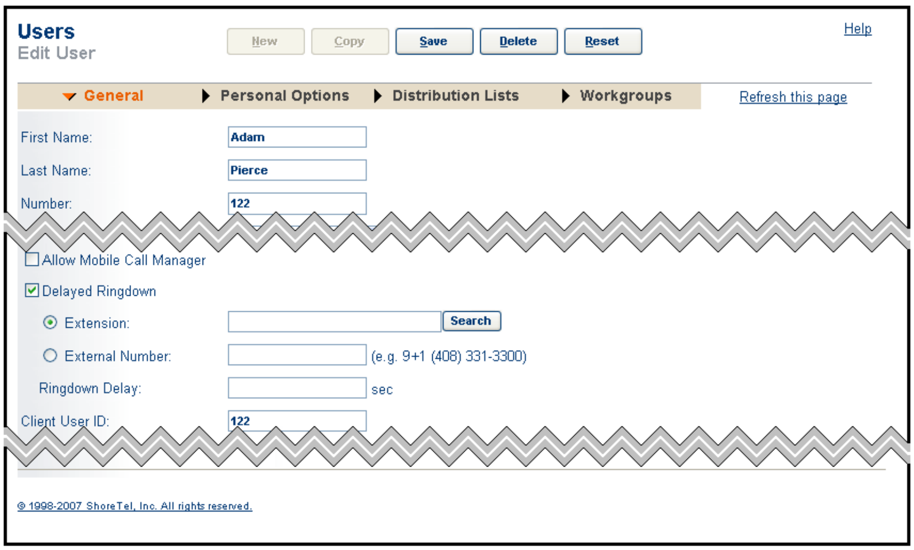


Figure 9-47 Delayed Ringdown Configuration Parameters

Delayed Ringdown Circuit parameters include:

- **Delayed Ringdown:** Select this checkbox to enable Delayed Ringdown for the extension. If this parameter is not enabled, all other Ringdown parameters are not selectable.
- **Extension:** Select this option to specify a ShoreTel extension as the recipient device, then enter the extension of the recipient device in the corresponding data entry field.
- **External Number:** Select this option to specify a phone outside of the ShoreTel network as the recipient device, then enter the phone number of the recipient device, including the Trunk Access Code of the ringdown trunk, in the corresponding data entry field.
- **Ringdown Delay:** Enter the period of time that the phone waits after the user picks up the handset before initiating the ringdown call. Enter 0 in this field to cause the phone to immediately dial the ringdown device whenever the handset is picked up.

9.15.3.2 Ringdown Procedures

The following sections describe the procedures required to implement various ringdown circuit configurations.

Implementing a One-to-One Unidirectional Ringdown Circuit

Unidirctional ringdown circuits require two Bridged Call Appearances – one for the calling device and one for the recipient device. Both devices in this procedure are ShoreTel extensions.

To implement a One-to-One Unidirectional Ringdown Circuit:

Step 1 Create two Bridged Call Appearances as follows:

Step a Launch ShoreTel Director.

Step b Click **Administration > Call Control > Bridged Call Appearances**.

Step c Click the NEW button. The Edit Bridged Call Appeances page appears.

Step d Set parameters for a new profile. See the "Configuring BCA Parameters" on page 246 for more information about creating a Bridged Call Appearance profile.

Step 2 Program an IP Phone Button on the calling device to make ringdown calls, as follows:

Step a Click **Administration > Users > Individual Users**. The Individual Users page appears.

Step b Select the user that you want to allow to use BCA. The Edit User page appears.

Step c Click the **Personal Options** tab.

Step d Click **Program IP Phone Buttons**. The Program IP Phone Buttons page appears.

Step e In the IP Phone Type field, select the type of phone the user uses for bridged calls.

Step f For the button that you want to program for the user, do the following:

- In the first Function field, select the **All** or **Telephony**.
- In the second Function field, select **Bridge Call Appearance**. The Program IP Phone Buttons page for the Bridged Call Appearance option appears as shown in Figure 9-48.

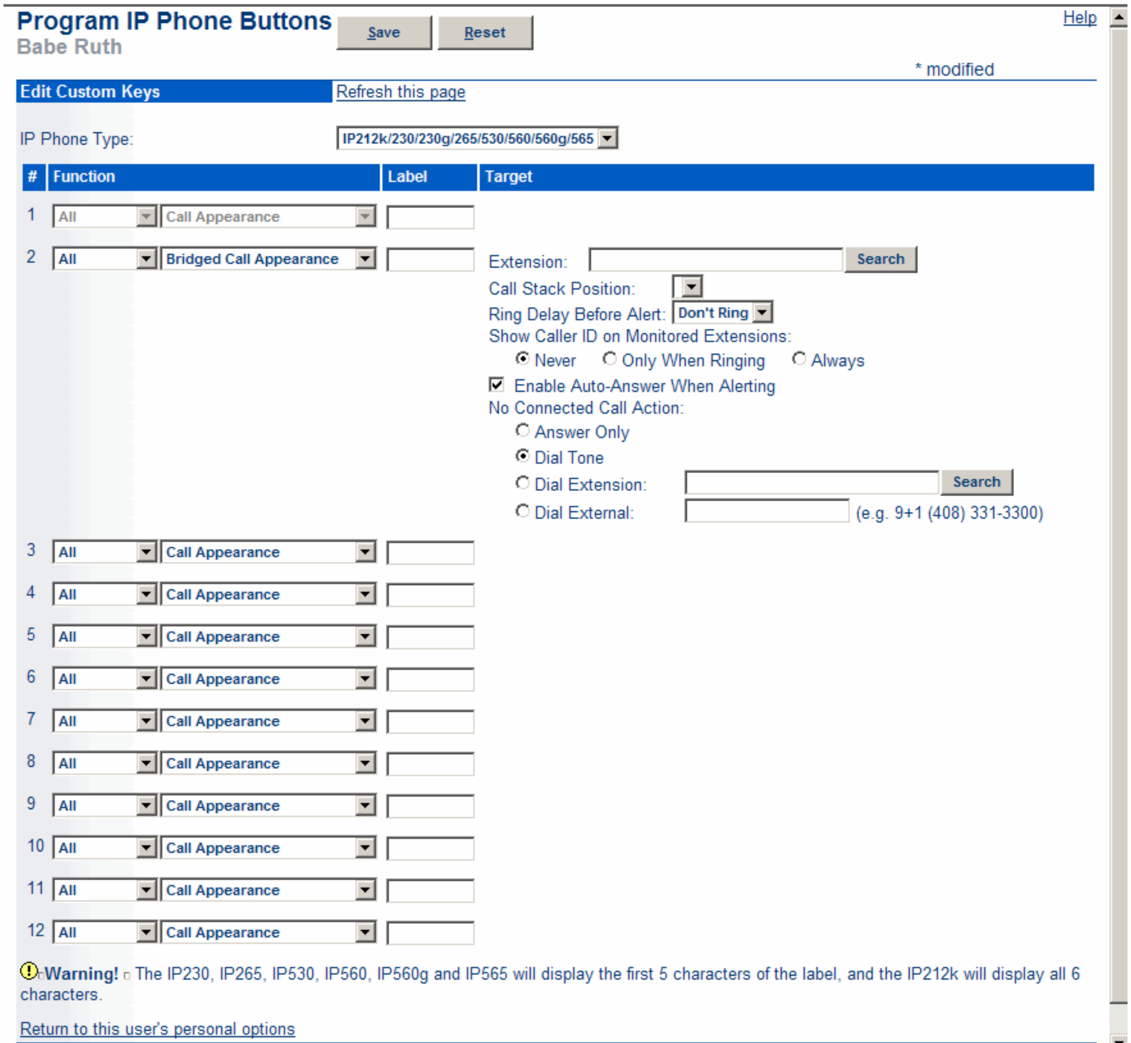


Figure 9-48 Program IP Phone Button: BCA Option Page

- In the Label field, enter a name that you want to assign this profile.
- In the Extension field, enter the extension number of the BCA profile with which you want to associate the user.
- In the Call Stack Position field, select the call escalation order with which you want to associate the user.
- In the Ring Delay Before Alert field, select the number of times that you want the phone of the user to ring before an automatic message is played.
- In the Show Caller ID on Monitored Extension field, click the radio button for the value that describes when you want the caller ID of this user to display for calls the user monitors.

- Check the **Enable Auto-Answere When Alerting** check box to the user to use standard means for answering phone calls.
- In the No Connected Call Action section, click the radio button for how the call is to be handle when there is not answer.

Step g Click **Save**.

Step 3 Program an IP Phone Button on the recipient device to receive ringdown calls, as follows:

Step a Create a button for a second user as described in Step 2 above.

Step b Enter the second Bridged Call Appearance extension number created in Step 1 in the Extension Field located right of the button function selection fields.

Step c Select **Dial Tone**, located below the Extension Field.

Step d Click **Save**.

Implementing a One-to-One Bidirectional Ringdown Circuit

Creating a bidirectional ringdown circuits differs from creating a unidirectional circuit in that the recipient device is configured to make ringdown calls. Both devices in this procedure are ShoreTel extensions.

To implement a One-to-One Bidirectional Ringdown Circuit:

Step 1 Create two Bridged Call Appearances.

Bridged Call Appearances are listed in the Bridged Call Appearance list page, which is accessed by selecting Administration -> Call Control -> Bridged Call Appearances from the Director menu. Press the New button at the top of this page to create new BCAs. Refer to the System Administrator's Guide for information on creating Bridged Call Appearance.

Step 2 Program an IP Phone Button on the calling device to make ringdown calls, as follows:

Step a Launch ShoreTel Director.

Step b Click **Administration > Users > Individual Users**.

Step c In the First Name column, click the name of the desired user. The Edit User page appears.

Step d Click the **Personal Options** tab.

Step e Click the **Program IP Phone Buttons** link. The Program IP Phone Buttons page appears.

Step f In the first Function field, select **All** or **Telephony**.

Step g In the second Function field, select **Bridged Call Appearance**. Bridged Call Appearance parameters populate the Target pane.

Step h Enter the first Bridged Call Appearance extension number created in Step 1 in the Extension Field located right of the button function selection fields.

Step i Select **Dial Tone**, located below the Extension field, and enter the second Bridged Call Appearance number created in Step 1 in the corresponding data entry field.

Step j Click **Save**.

Step 3 Program an IP Phone Button on the recipient device to make ringdown calls, as follows:

Step a Repeat Step a through Step e (under Step 2) for the recipient user's extension.

Step b Enter the second Bridged Call Appearance extension number created in Step 1 in the Extension Field located right of the button function selection fields.

Step c Select **Dial Tone**, located below the Extension field, and enter the first Bridged Call Appearance number created in Step 1 in the corresponding data entry field.

Step d Press **Save**.

Creating One-to-Many Ringdown Circuits

The process for creating a One-to-Many Ringdown Circuit differs from creating a One-to-One circuit in that the process of configuring the recipient devices is repeated for each recipient device in the ringdown network.

Implementing a Delayed Ringdown Circuit

A unidirectional delayed ringdown circuit requires one Bridged Call Appearance that is assigned to the recipient end of the circuit.

To create a unidirectional delayed ringdown circuit:

Step 1 Create one Bridged Call Appearance.

Step 2 Program a phone to make ringdown calls, as follows:

Step a Open the Individual User list page in Director by selecting **Administration -> Users -> Individual Users** from the main menu.

Step b Open the Edit User page by clicking the First Name of the desired user in the Individual User List.

Step c Click the General tab.

Step d Select **Delayed Ringdown** and do one of the following:

- Click the **Extension** radio button and enter the Bridged Call Appearance extension that you want to associate with the user in the field.
- Click the External Number radio button and enter the external number that you want to associate the user with in the field.

Step e In the Ringdown Delay field, enter period in seconds that you want the system to wait for the receiver to respond before dialing the ringdown number.

The delay period is the time between when the handset is lifted and when the ringdown call is initiated.

Step 3 Program an IP Phone Button on the recipient device to receive ringdown calls, as follows:

Step a Open the Individual User list page in Director by selecting **Administration -> Users -> Individual Users** from the main menu.

Step b Open the Edit User page by clicking the First Name of the desired user in the Individual User List.

Step c Select **Personal Options** on the page selection bar at the top of the page.

Step d Click **Program IP Phone Buttons** to open the Program IP Phone Buttons page.

Step e Select **Bridged Call Appearance** as the function for the desired IP button.

Step f Enter the initiator's extension number in the Extension Field located right of the button function selection fields.

Step g Select **Dial Tone**, located below the Extension Field.

Step h Click **Save**.

Creating a Ringdown Circuit with an External Endpoint

The process for creating a ringdown circuit to an external endpoint differs from the procedure for circuits with internal endpoint as follows:

- The number of the recipient device is entered in External Number data entry fields instead of the Extension data fields.

  These fields are located in the Program IP Phone Buttons page and the Edit User page.

- The external number must be accessed through a specific trunk that is configured as the only trunk within a trunk group.
- The number of the recipient device includes the Trunk Access Code of the specified trunk group.

9.16 Media Encryption

ShoreTel encrypts RTP (payload) packets within the ShoreTel network. Call control packets are not encrypted.

9.16.1 System Support

Encryption is enabled or disabled through ShoreTel Director by a system administrator. Encryption is enabled or disabled only on a system basis and cannot be enabled for individual devices or selected calls. Encryption is transparent to the end user and end users have no control over which calls are encrypted. Changing the system encryption setting does not alter calls that are in progress; unencrypted calls in place when encryption is enabled remain unencrypted until the calls are terminated.

System administrators enable and select an encryption algorithm through ShoreTel Director. The following encryption options are available:

- None
- 128 bit ShoreTel Proprietary

- SRTP - 128 bit AES

SRTP with AES and authentication has a significant impact on the system load when a large number of media channels are encrypted.

SRTP-AES encryption is provided on all ShoreTel 9 and later systems and does not require an additional license.

9.16.2 Devices

9.16.2.1 Switches and Codecs

Encryption is supported by the following ShoreTel Voice Switches:

- All Voicemail Model switches
- All 1-U Half Width switches
- All 1-U Full Width switches

Switches do not support SRTP with linear (LRNB/8000) or wide-band (LRWB/16000) codecs. When SRTP is enabled, codec negotiation excludes these codecs.

ShoreTel switches support a maximum of 36 encrypted media streams. This limitation potentially impacts switches that provide T1 or E1 channels with high 3-way conference call traffic.

Each channel in a 3-way conference requires two media stream encryption resources, limiting switches to 18 encrypted channels for 3-way conferences. In this scenario, all remaining trunks provided by the switch are blocked while 18 channels are engaged in 3-way conference calls. Switches can service any combination of two-way (one encrypted media stream) and three-way (two encrypted media streams) calls that do not exceed 36 media streams. Analog ports on the SG-220T1A are included in this limitation.

9.16.2.2 Phones and Applications

All ShoreTel IP Phones that run support encryption. SoftPhone, which is available through Communicator, also supports encryption.

Communicator and the ShoreTel IP Phones display a padlock icon for each call utilizing SRTP encryption is active. The Call History list also uses the padlock icon to denote encrypted calls. ShoreTel IP Phones IP210, IP110, and IP115 do not display the encrypted call padlock icon. The padlock icon indicates the call media is secure on the ShoreTel network; ShoreTel cannot guarantee call security outside of the network, such as calls that terminate across an analog or digital trunk. The padlock icon is not displayed when Proprietary Encryption is active.

The ShoreTel Conference Bridge, the IP 8000 Conference Phone, voicemail and auto attendant do not support SRTP-AES encryption.

Phones that do not support SRTP cannot perform barge in, whisper, or silent monitor functions on existing calls that are SRTP encrypted. When added to a call using SRTP, new parties using devices that do not support SRTP exchange unencrypted media streams. SRTP does not address user registration, call setup, or signaling related security.

9.16.3 Implementation

The Enable Media Encryption parameter on the Call Control Options page, controls encryption.

To configure encryption:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Call Control > Options**. The Call Control Options page appears as shown in Figure 9-49.

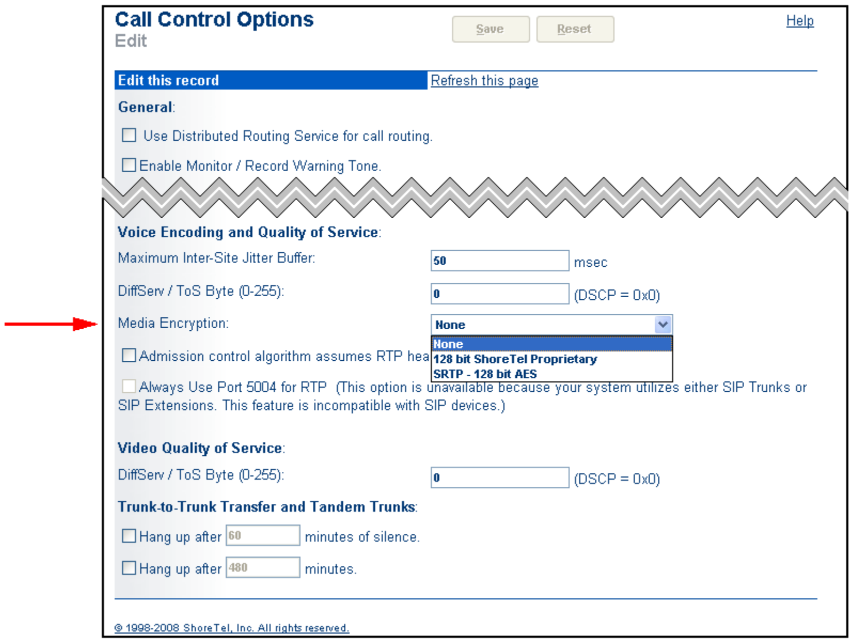


Figure 9-49 **Enable Media Encryption option on the Call Control Options Page**

Step 3 In the Media Encryption field, select the desired option.

Step 4 Click **Save**.

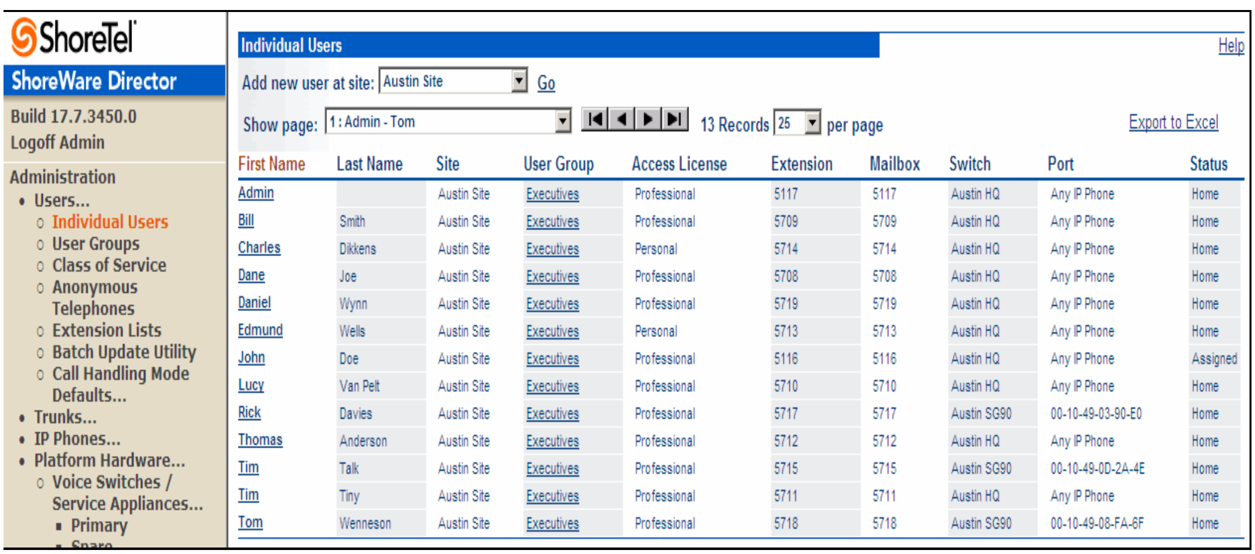
C H A P T E R 1 0

Configuring Users

This chapter provides information about configuring user parameters. The topics discussed include:

10.1 Overview

The Users link in the Director navigation pane provides access to the windows for Individual Users, User Groups, Class of Service, Notify Users, Anonymous Telephones, Extension Lists, Batch Update Utility, and Call Handling Mode Defaults. Within these windows, user accounts can be added and edited and receive assignments for their phone properties. Figure 10-1 shows the navigation pane and the *Individual Users* list.



First Name	Last Name	Site	User Group	Access License	Extension	Mailbox	Switch	Port	Status
Admin		Austin Site	Executives	Professional	5117	5117	Austin HQ	Any IP Phone	Home
Bill	Smith	Austin Site	Executives	Professional	5709	5709	Austin HQ	Any IP Phone	Home
Charles	Dikkens	Austin Site	Executives	Personal	5714	5714	Austin HQ	Any IP Phone	Home
Dane	Joe	Austin Site	Executives	Professional	5708	5708	Austin HQ	Any IP Phone	Home
Daniel	Wynn	Austin Site	Executives	Professional	5719	5719	Austin HQ	Any IP Phone	Home
Edmund	Wells	Austin Site	Executives	Personal	5713	5713	Austin HQ	Any IP Phone	Home
John	Doe	Austin Site	Executives	Professional	5116	5116	Austin HQ	Any IP Phone	Assigned
Lucy	Van Pelt	Austin Site	Executives	Professional	5710	5710	Austin HQ	Any IP Phone	Home
Rick	Davies	Austin Site	Executives	Professional	5717	5717	Austin SG90	00-10-49-03-90-E0	Home
Thomas	Anderson	Austin Site	Executives	Professional	5712	5712	Austin HQ	Any IP Phone	Home
Tim	Talk	Austin Site	Executives	Professional	5715	5715	Austin SG90	00-10-49-0D-2A-4E	Home
Tim	Tiny	Austin Site	Executives	Professional	5711	5711	Austin HQ	Any IP Phone	Home
Tom	Wenneson	Austin Site	Executives	Professional	5718	5718	Austin SG90	00-10-49-08-FA-6F	Home

Figure 10-1 Users Navigation Links

The task of creating user accounts actually begins with the specification of the classes of service (COS). In the larger sense, configuring users has a specific order:

1. Classes of service (COS)
2. User groups
3. Individual users

Having the following information expedites the whole user configuration process:

- A list or outline of the COSs that are appropriate for the ShoreTel deployment and available for assignment to user groups
- A list or outline of user groups to which individual users are assigned
- A list of new users to add to the system

A ShoreTel system comes with default classes of service and user groups.

10.2 Classes of Service

A class of service grants permission to a specified set of features and privileges. Users assigned to a class of service can access the specified features and privileges. ShoreTel defines three types of service classes: telephony features, call permissions, and voicemail permissions.

Configuring classes of service is the first step in configuring users. To configure Class of Service, you must access the Class of Service page. To access the Class of Service page, do the following:

Step 1 Launch **ShoreTel Director**.

Step 2 Click **Administration > Users > Class of Service**. The Class of Service page appears as shown in Figure 10-2.

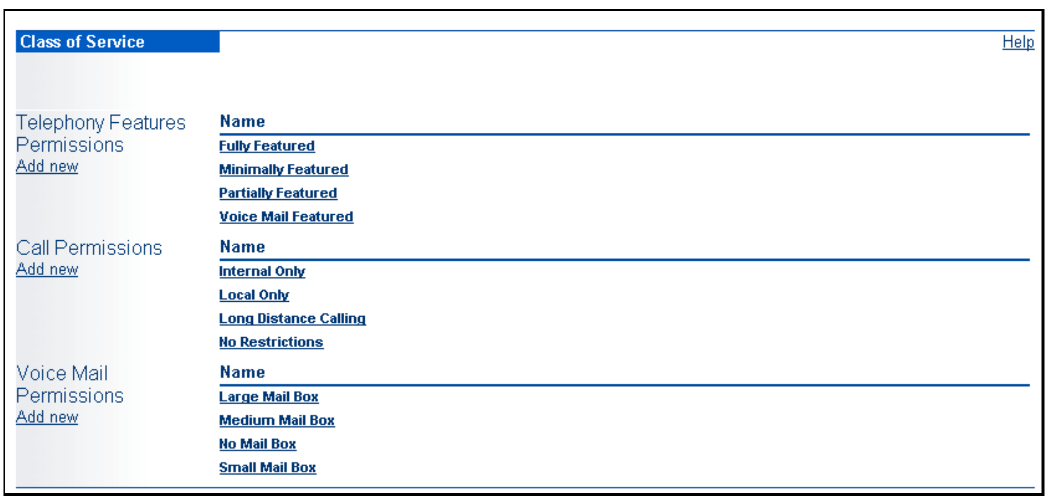


Figure 10-2 Class of Service Page

You can configure permissions for telephony features, outside calling, and voice mail usage. Descriptions and instructions are provided for these permissions in the following sections:

- "Telephony Features Permissions" on page 315

- "Call Permissions" on page 321
- "Voice Mail Permissions" on page 323

10.2.1 Telephony Features Permissions

This section describes the classes of service that can be configured from the Telephony Features Permissions edit page. Telephony features permissions are assigned to user groups and define how users can use their telephone features, such as call stack depth, paging, and call forwarding to an external number.

To configure a telephony feature class of service:

Step 1 Navigate to the Class of Service window.

Step 2 Click one of the preconfigured COS profiles (Fully Featured, Minimally Featured, or Partially Featured) or the Add New link to create a new class of service for Telephony Features. The Edit Telephony Features Permissions page for the class of service you select appears as shown in Figure 10-3.

Figure 10-3 Telephony Features Permissions Edit Page – Top

Step 3 Specify the following COS parameters:

- **Name:** This specifies a descriptive name of the COS that you are creating or editing.
- **Max. Call Stack Depth:** This lets you specify the maximum number of simultaneous calls that can be "stacked" on a user's extension. When this number is met, additional inbound calls are routed to the call forward busy destination.

  Valid entries are 1 through 16
- **Max Buddies Per Users:** This option specifies the number of individuals that a service class member can designate as a contact in Communicator. Users can monitor presence status of their contacts.
- **Max. Parties in Make Me Conference:** This lets you specify the maximum number of parties that can be included in any Make Me Conference call made from your site. Select the number of parties from the drop-down. The default number is 3. Valid values are 3 through 6.
- **IM Presence Invitation Handling:** This option designates the default method of handling requests for viewing IM presence status. Each user can specify personal handling methods through Communicator.
- **Allow Call Pickup:** This check box enables call pickup, which lets users pick up any ringing extension (including the night bell) or pick up any parked call.
- **Allow Trunk-to-Trunk Transfer:** This check box enables trunk-to-trunk call transfers. A trunk-to-trunk transfer is defined as follows:
  - — An internal party is talking with an external party. The internal party transfers the external party *blindly* or *consultatively* to an external party via the telephone or a ShoreTel Communicator application.
  - — A three-party conference call with one internal party and two external parties when the internal party drops from the call via the telephone or a ShoreTel Communicator application.

  A trunk-to-trunk transfer is not an external party forwarded to an external number by a user's call handling mode. This is not to be confused with **Allow External Call-forwarding**.

  This feature is automatic and by-passes toll-related call permissions. If there is the potential for toll fraud by employees who might abuse this feature, limit this permission to only a few select user groups. For example, only the executive and sales user groups may be granted permission.

  If this parameter is enabled, trunk-to-trunk transfers can be managed from the Call Control Options edit page.
- **Allow Overhead and Group Paging:** This check box lets users dial any site paging extension and make an announcement using the overhead paging system or group paging.
- **Allow Make Hunt Group Busy:** If granted, this permission allows the user to busy out a hunt group or return it to service by keying *18 on the telephone keypad. In addition, the Quick Look Maintenance page can be used to busy out or return a hunt group to service. Calls will be forwarded to the Busy Destination set in the Hunt Groups page.

  Should a hunt group be busied out while on a holiday or off-hours schedule, the schedule takes precedence.

- **Allow Extension Reassignment:** This check box lets users reassign their extension to another telephone.
- **Allow PSTN Failover:** This check box indicates that site-to-site calls that fail over proprietary routes will be automatically made over a PSTN number. The PSTN number to use in case of failure must be entered on the User edit page.
- **Show Caller ID Name and Number on Monitored Extensions:** Select this check box to enable the monitoring party to see information about the incoming calls on the monitored party's line. If unchecked, the incoming call will still be shown with an icon and LED, but not textually in the announcement area.
- **Enumerate Individual Held Calls for Unpark:** Select this option to authorize users to view individual calls parked on another user's call stack. User's require this authorize to specify the call to be unparked from another user's stack.
- **Allow Customization of IP Phone Buttons and Communicator Monitor Windows:** Select this check box to allow users to configure the programmable buttons on their ShoreTel IP Phone or BB24 device.

  Clear the check box to prevent users from being able to configure custom buttons. This action will, for example, prevent users from being able to configure their phones to monitor the extension of anyone they please and will require that extension monitor, and other features, be set up by a system administrator.

  Without the box checked, only the system administrator can configure a user's phone for extension monitoring. Extension Monitor will only monitor those extensions that are monitored on the phones. If there are none, extension Monitor is disabled.
- **Show Extensions with Different Prefixes in Directory:** Select this check box to display extensions that have different prefixes. The On-Net Dialing feature often results in a remote site having a different prefix from a headquarters site. Enabling this check box will cause all extensions (even those at the remote sites) to appear in the directory.
- **Allow Collaboration Features:** Checking this box allows document sharing. See Chapter 11: Configuring Client Applications on page 361 for additional information
- **Allow Recording of Own Calls:** Checking this box grants permission to record one's own calls. This will also be impacted by the Enable Monitor/Record Warning Tone parameter in Chapter 9.
- **Allow Intersite Video Calls:** When this option is selected, class members are authorized for video calls with users located at other ShoreTel sites.
- **Allow Call Notes:** Enables the Call Notes feature in ShoreTel Communicator that allows users to make text notes during calls. Notes appear in the call history.

  NOTE If the Call Notes feature is disabled in ShoreTel Director, existing notes will remain available for users to view but will not be editable.
- **Show Call History:** Enables the Call History feature in ShoreTel Communicator. This feature allows users to view call activity on their ShoreTel extension.
- **Directed Intercom/Paging:** The Allow **initiation** check box and Accept radio buttons can be used to configure the intercom/paging calls function.

  For more information about this feature, see the "Intercom, Whisper Paging, Barge In, Record, and Monitor" on page 436.

- **Barge In**: Barge In permits one party to join an existing call as a fully conferenced participant. When barge in is initiated, a brief intrusion tone is played to the other participants and (if present) the monitor/record warning tone is discontinued. Checking **Allow Initiation** allows users with this CoS to barge in on users with **Accept** Barge In enabled.

  See the "Call Handling Mode Delegation" on page 435 for details on configuring this option.

- **Record Other's Calls:** Select this check box to allow a user to record the calls of another user (for example, a that a supervisor can record the calls of an agent in a call center).

  The "Selectable Mailboxes" feature allows the automatic placement of recorded calls into mailboxes other than that of the client who initiated the call recording. See "Record Call" and "Record Extension" in Table 12-1 for details.

  A monitor/record warning tone is played. To make recordings silent, disable the warning tone. See the"General Parameters Area in the Options Window" on page 291 for details.

  See the "Intercom, Whisper Paging, Barge In, Record, and Monitor" on page 436 for details on configuring this option.

- **Silent Monitor/Silent Coach Other's Calls:** Allows a supervisor to monitor a phone call of a user and to speak to the user without the other party hearing. These check boxes allow the user to monitor calls. The monitor/record warning tone is played if enabled in Call Control.

  See the "Intercom, Whisper Paging, Barge In, Record, and Monitor" on page 436 for details on configuring this option.

- **Allow Call Handling Changes:** These check boxes allows users to make changes to their call handling settings.

  — **Current Mode—**Checking this box allows users to change their current call handling mode from ShoreTel IP Phone IP phones and Personal Communicator.

  — **Detailed Settings—**Checking this box allows users to change all call handling settings, such as call forward destinations, from Personal Communicator.

- **Allow External Call-forwarding and Find Me Destinations:** This check box lets users forward incoming calls to an external number, as restricted by the Scope, Restrictions, and Exceptions parameters.

  This must not be confused with Allow Call Transfer Trunk to Trunk.

- **Allow External Extension Reassignment:** This check box allows users in this COS to assign their extension to a PSTN phone for use with the Extension Assignment feature. If this check box is marked, the user is able to use a cell phone or home phone as an extension of a phone in the ShoreTel network.

  — **Allows External Assignment:**

  Allow Additional Phones to Ring Simultaneously and to Move Calls: This feature lets users configure one or two additional phones to ring at the same time as their main ShoreTel phone rings. The user can specify the phone number of additional phones in ShoreTel Communicator, or the administrator can configure these phone numbers in the ShoreTel Director's individual user page (Personal Options > External Assignment and Additional Phones).

After this feature is configured, the user simultaneously receives calls on the ShoreTel extension and his or her additional phones. As a convenience, the user can suspend this function by using Communicator. Also, an optional ring delay can be configured by the Administrator to let the main telephone ring a configurable number of times before the additional phones start ringing.

After the Simultaneous Ringing call is established, the call can be moved between the Simultaneous Ringing devices. The Call Move mechanism can be initiated by a “Call Move” IP Phone programmable button on the ShoreTel IP 655 Phone (other options are available on ShoreTel IP Phones, Analog and SIP devices, and ShoreTel Communicator).

- **Scope:** Scope allows a general permission level setting. Levels are presented from the most restrictive to the most permissive. The Restrictions and Permissions listed are applied in addition to the general scope setting.
  - **Local Only** allows forwarding or extension reassignment only to local or additional local area codes, as defined on the Site edit page.
  - **National Long Distance** also allows forwarding or extension reassignment to long-distance numbers within the country, as defined on the Site edit page.
  - **National Mobile** allows forwarding or extension reassignment to mobile numbers. Some countries use “caller pays” mobile calling, so do not select this radio to avoid incurring the associated costs of calls being sent to a mobile phone.
  - **International Long Distance** also allows forwarding or extension reassignment to international numbers, as defined on the Site edit page.
  - **All Calls** allows forwarding or extension reassignment to any number, including 900, Operator Assisted, and Carrier Select numbers. This is the default.
- **Restrictions:** Restrictions are applied in addition to the Scope. Follow these rules for specifying restrictions:
  - The comma-separated restriction expressions have a limit of 50 characters, total (including commas and semicolons).
  - Numbers must be entered in canonical format, including the international designation “+” and country code. For example, to restrict forwarding to the 408 area code in the U.S., use +1408.
  - Non-routable calls (311, 411, etc.) for a country must be designated by the country code plus the “/” character. For example, to restrict forwarding to 311 in the U.S., use 1/311.
  - Each field can contain multiple entries as long as they are separated by commas or semicolons.
  - Each entry must consist of numbers only.
  - Access codes (such as 9) must not be included
  - The wildcard of “x” can be used.

  When a call is both restricted and permitted, it is permitted. For example, restricting +1 408 and permitting +1 408 331 restricts all calls to the 408 area code except those to 408 331-xxxx.

- **Permissions:** Permissions are applied in addition to the Scope. Follow these guidelines for specifying permissions:
  - — The comma-separated permission expressions have a limit of 50 characters, total (including commas and semicolons).
  - — Numbers must be entered in canonical format including the international designation "+" and country code. For example, to permit forwarding to the 408331 area code and prefix in the U.S., use +1408331.
  - — Non-routable calls (311, 411, etc.) for a country must be designated by the country code plus the "/" character. For example, to permit forwarding to 311 in the U.S., use 1/311.
  - — Each field can contain multiple entries as long as they are separated by commas or semicolons.
  - — Each entry must consist of numbers only.
  - — Access codes, such as 9, must not be included.
  - — The wildcard of "x" can be used.

  When a call is both restricted and permitted, it is permitted. For example, restricting +1 408 and permitting +1 408 331 restricts all calls to the 408 area code except those to 408 331-xxxx.

10.2.2 Call Permissions

This section describes the types of call permissions the ShoreTel administrator can set. Call permissions are classes of service that specify the type of call users are allowed to dial. Call permissions are assigned to user groups. Figure 10-4 shows the Call Permissions edit page.

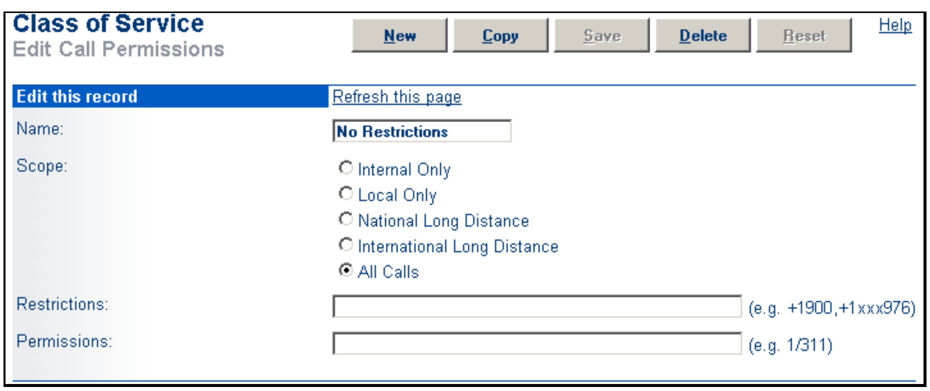


Figure 10-4 Call Permissions Edit Page

The parameters on the Call Permissions edit page are as follows:

- **Name:** A descriptive name of the COS being added or edited.
- **Scope:** The scope is a general permission level. The scopes in the window descend from most restrictive to most permissive. The Restrictions and Permissions list is applied in addition to the general scope for the COS.
  - — **Internal Only** allows calls only to internal extensions and to the configured emergency number.

— **Local Only** allows calls only to local or additional local area codes, as defined on the Site edit page.

— **National Long Distance** also allows calls to long-distance numbers within the country, as defined on the ***Site*** edit page.

— **International Long Distance** also allows calls to international numbers, as defined on the ***Site*** edit page.

— **All Calls** allows calls to any number, including 900, Operator Assisted, and Carrier Select numbers, as well as use of Vertical Service Codes. This is the default.

- **Restrictions:** Restrictions are applied in addition to the Scope setting. The rules for specifying restrictions are as follows:

  — For this COS, the maximum number of characters in the comma-separated restriction expressions is 255.

  — Numbers must be entered in canonical format including the international designation "+" and country code. For example, to restrict calls to the 408 area code in the U.S., type +1408.

  — Non-routable calls (311, 411, and so on) for a country must be designated by the country code plus the "/" character. For example, to restrict 311 in the U.S., type 1/311

  — Each field can contain multiple entries as long as they are separated by commas or semicolons.

  — Each entry must consist of numbers only.

  — Access codes, such as 9, must not be included.

  — The wildcard of "x" can be used.

  If a call is both restricted and permitted, it is permitted. For example, restricting +1 408 and permitting +1 408 331 restricts all calls to the 408 area code except those to 408 331-xxxx.

- **Permissions:** Permission are applied in addition to the Scope setting. The rules for entering restrictions are as follows:

  — For this COS, the maximum number of characters in the comma-separated permission expressions is 255.

  — In general, numbers must be entered in canonical format including the international designation "+" and country code. For example, to permit calls to the 408331 area code and prefix in the U.S., use +1408331.

  — Non-routable calls (311, 411, and so on) for a country must be designated by the country code plus the "/" character. For example, to permit 311 in the U.S., use 1/311

  — Each field can contain multiple entries as long as they are separated by commas or semicolons.

  — Each entry must consist of numbers only.

  — Access codes, such as 9, must not be included.

  — The wildcard of "x" can be used.

If a call is both restricted and permitted, it is permitted. For example, restricting +1 408 and permitting +1 408 331 restricts all calls to the 408 area code except those to 408 331-xxxx.

10.2.3 Voice Mail Permissions

This section describes the classes of service that you configure from the Voice Mail Permissions edit page, shown in Figure 10-5. Voice mail permissions are assigned to user groups, providing users with specific usage of the ShoreTel voice mail system.

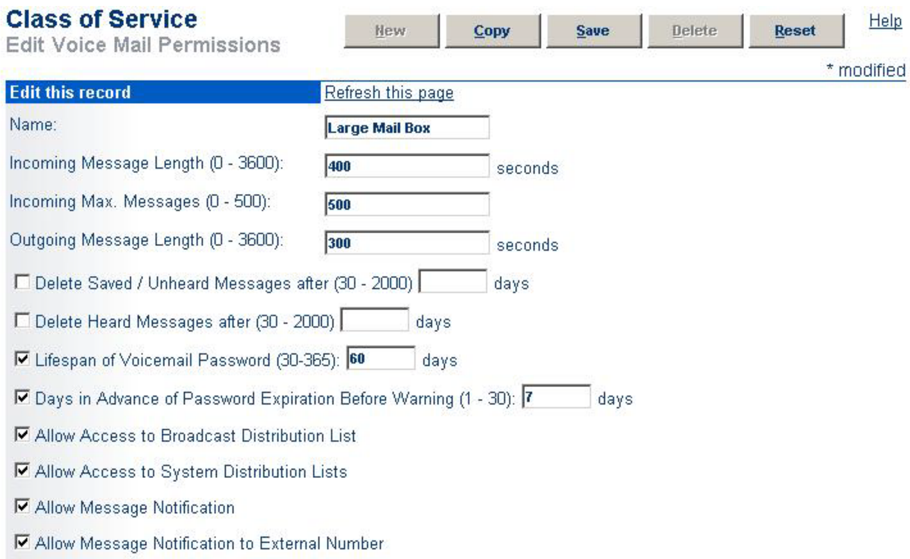


Figure 10-5 Voice Mail Permissions Edit Page

The parameters on the Voice Mail Permissions edit page are as follows:

- **Name:** This is a descriptive name of the CoS record that you are creating or editing.
- **Incoming Message Length:** This is the maximum length of an incoming voice mail message.

  Valid values are from 1 to 3600 seconds. The default is 300 seconds.
- **Incoming Max. Messages:** This is the maximum number of messages that can be queued in a mailbox, including new, saved, and deleted messages.

  Valid values are from 1 to 500 messages.
- **Outgoing Message Length:** This is the maximum message length that a user can record before sending a message to another extension. This parameter controls both the composed message and the greeting.

  Do not confuse this with the user's personal voice mail greeting.

  Valid values are from 1 to 3600 seconds.
- **Lifespan of Voicemail Password:** Select this check box to have the system periodically require users to change their TUI passwords. It is recommended to enable this feature as requiring periodic password changes increases the security of the system.

  The password change applies to the following Dialed Number types:

— User extensions

— Workgroup extensions

— Route point extensions

— External user extension

Valid values are from 30 to 365 days. The default value for the time limit is 90 days.

- **Days in Advance of Password Expiration Before Warning:** When the Lifespan of Voicemail Password is selected, this option to warn users that their password is about to expire also becomes active. Select the check box to enable the feature, thus allowing users to be notified prior to the expiration of their passwords, and giving them time to proactively change their passwords days or weeks in advance of the actual expiration date.

  Enable the feature by selecting the check box and then entering the number of days ((prior to the password expiration), at which point group members will begin receiving warning messages. If the check box is not selected or if "0" is placed in the data entry field, password expiration warning messages are not delivered to the group member.

  Valid values are from 1 to 30 days. The default value for the time limit is 7 days.

- **Allow Access to Broadcast Distribution List:** This gives users access to the company-wide distribution list. A user with this permission is able to broadcast voice mail messages to all users throughout the company.

- **Allow Access to System Distribution Lists:** This gives users access to system distribution lists.

- **Allow Message Notification:** This enables message notification. It might be further qualified by Allow Message Notification to External Number.

  This is enabled as a system default.

- **Allow Message Notification to External Number:** This permits message notification to an external number. It cannot be set unless Allow Message Notificatio*n* is enabled.

  This is enabled as a system default.

- **Auto-Delete:**

  — **Delete Saved / Unheard Messages after (30-2000) days:** Select this check box to have the system automatically delete saved or unheard voice mail messages that are older than *n* days.

  Valid values are from 30 to 2000 days. The default value for the time limit is 0 days (meaning the feature is disabled).

  — **Delete Heard Messages after (30-2000) days:** Select this check box to have the system automatically delete heard voice mail messages after *n* days.

  Valid values are from 30 to 2000 days. The default value for the time limit is 0 days (meaning the feature is disabled).

  As with Saved/Unheard messages, the user will receive several warnings before messages are deleted (see above for details).

  — **Enable Auto-Delete Notification:** Select this check box to have the system automatically delete heard voice mail messages after *n* days. This option is available only if "Delete Saved / Unheard Messages after (30-2000) days" or "Delete Heard Messages after (30-2000) days" is selected.

If enabled, users who have messages older than the expiry time limit will receive multiple warnings in the form of email and/or voice mail messages indicating that their voice mail messages (that exceed the time limit) will be deleted. The first warning is sent two weeks before the expiration deadline, and a second message is sent one week prior to deadline.

After the time limit has passed, the user will receive a message similar to the following: “Your mailbox was cleaned automatically; n messages were deleted.”

This message can be emailed to the user if an email address is specified in the profile of the ShoreTel recipient.

10.3 User Groups

Configuring user groups is the second step in configuring users.

10.3.1 Viewing User Groups

To view the available user groups, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Users > User Groups**. The User Groups page appears as shown in Figure 10-6.

User Groups

Help

Add new

Name	Telephony Features (Permissions)	Call (Permissions)	Voice Mail (Permissions)	SMDI	Account Codes	DID As CESID
Account Code Service	Minimally Featured	No Restrictions	No Mail Box	None	Disabled	Yes
Anonymous Telephones	Minimally Featured	Internal Only	No Mail Box	None	Disabled	Yes
Executives	Fully Featured	No Restrictions	Large Mail Box	None	Disabled	Yes
House Telephones	Minimally Featured	Internal Only	No Mail Box	None	Disabled	Yes
IP Telephones	Minimally Featured	Internal Only	No Mail Box	None	Disabled	Yes
Managers	Partially Featured	No Restrictions	Medium Mail Box	None	Disabled	Yes
Staff	Partially Featured	Long Distance Calling	Medium Mail Box	None	Disabled	Yes
Voice Mail Notification	Voice Mail Featured	No Restrictions	No Mail Box	None	Disabled	Yes

Figure 10-6 User Groups List Page

User Group List page column definitions are as follows:

- **Name:** This is the name of the user group.
- **Telephony Features:** This is the telephony features permissions CoS associated with the user group.
- **Call:** This is the call permissions CoS associated with the user group.
- **Voice Mail:** This is the voice mail CoS associated with the user group.
- **SMDI:** This is the Simplified Message Desk Interface mode set for the user group. Also see the "Legacy Voice Mail Integration" on page 75.
- **Account Codes:** This is the account code collection mode set for the user group. Also see the "Multi-site Account Codes" on page 240.

- **DID As CESID:** Indicates whether a DID number should be sent as the Caller's Emergency Service ID number for a 911 emergency call.

10.3.2 Creating a User Group

To create a user, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Users > User Groups**. The User Groups page appears.

Step 3 Click **Add new**. The Edit User Group page appears as shown in Figure 10-7.

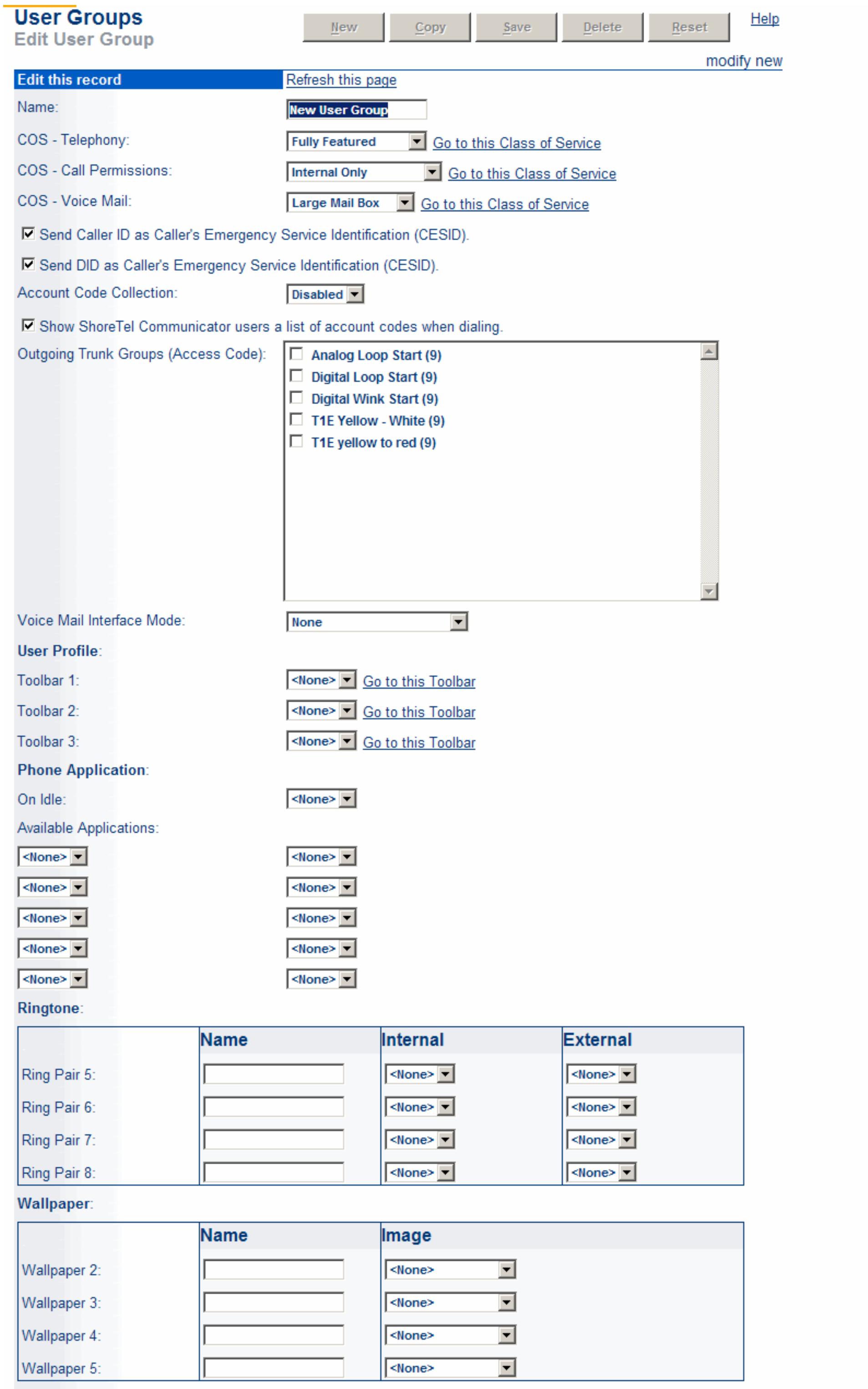


Figure 10-7 User Group Edit Page

Step 4 Configure parameters for the user group as follows:

- **Name:** This is a descriptive name of the group you are adding or editing.

- **COS – Telephony:** This is the telephony features permissions COS associated with the user group.
- **COS – Call Permissions:** This call permission COS is associated with a user group.
- **COS – Voice Mail:** This is the voice mail permissions COS record associated with the user group.
- **Send Caller ID as Caller's Emergency Service Identification (CESID):** The caller ID configured on the User's page is the telephone number sent to the service provider when a user dials an emergency services number (e.g., 911 in the U.S.). Default setting is checked on. If this option is not selected, the outbound caller ID will be either the user's DID or the site's CESID.

  For more information on setting up emergency dialing, see Appendix A, "Emergency Dialing Operations" for more information.
- **Send DID as Caller's Emergency Service Identification (CESID):** This telephone number is sent to the service provider when a user dials an emergency services number from a home phone (for example, 911 in the U.S.). If this option is not selected and Send Caller ID as Caller's Emergency Service Identification (CESID) is also not selected, the outbound caller ID becomes the site's CESID.

  For more information on emergency and 911 calls, see Appendix A, "Emergency Dialing Operations" for more information.
- **Account Code Collection:** Select from Disabled, Optional, or Forced account code collection for the selected user group.
  - **Disabled—**Account collection is not active for this group.
  - **Optional—**Users are prompted to enter an account code. If no account code is entered, the call is completed without account code records.
  - **Forced—**Users must enter an account code for all calls outside the bounds of the call permissions set for the user.

  For more information about account codes, see the "Multi-site Account Codes" on page 240.
- **Show Communicator users a list of account codes when dialing:** Enabling this option allows Communicator users to select an account code from the complete list of account codes when prompted for an account code. Disable this feature to restrict the user's visibility of account codes.
- **Outgoing Trunk Groups (Access Code):** Select the trunk groups to which this user group has access for outgoing calls. You can assign multiple trunk groups for this user group.
- Voice Mail Interface Mode: This parameter allows you to specify special routing to the voice mailbox for users associated with this user group. Choose any of the following:
  - **None:** The user uses a standard ShoreTel voice mailbox.
  - **External Voice Mail, QSIG:** Connects to an external voicemail system using QSIG protocol.
  - **External Voice Mail, SMDI:** Connects to an external voicemail system using SMDI protocol.
  - **External Voice Mail, SIP:** Connects to an external voicemail system using SIP protocol.

— **ShoreTel Voice Mail, SMDI:** Connects to a legacy ShoreTel system using SMDI protocol.

- User Profile:
- Phone Application:
  — On Idle:
  — Available Applications:
- Ringtone:
- Wallpaper:

10.4 Individual Users

Configuring individual users is the final step in setting up user parameters. From the Individual Users list page, shown in Figure 10-8, invoke the Edit User page by selecting a site and clicking Go. The Edit User page specifies the user's general information, personal options, distribution lists, and workgroups.

First Name	Last Name	Site	User Group	Client Type	Extension	Mailbox	Switch	Port	Status
Adam	Ant	Headquarters	Executives	Operator	4509	4509	Doc-24	ShoreDoc IP Phone	Home
Betty	Boop	Headquarters	Executives	Operator	4510	4510	Doc-24	10	Home
Carl	Castle	Headquarters	Executives	Operator	4511	4511	Doc-24	11	Home
Donald	Duck	Headquarters	Executives	Operator	4512	4512	ShoreDoc-HQ	Any IP Phone	Home
Ernie	Els	Headquarters	Executives	Personal	4513	4513	Doc-24	13	Home
Fred	Flintstone	Headquarters	Executives	Personal	4514	4514	Doc-24	14	Home
Greta	Garbo	Headquarters	Executives	Personal	4515	4515	Doc-24	15	Home
Howard	Hughes	Headquarters	Executives	Personal	4516	4516	Doc-24	16	Home
Johnathan	Seagull	Headquarters	Executives	Personal	4805	4805	Doc-24	1	Home
Tim	Toolman	Headquarters	Executives	Personal	4904	4904	Bedrock	10	Home
Ursula	Urp	Headquarters	Executives	Personal	4907	4907	Bedrock	13	Home
Vince	Vinceble	Headquarters	Executives	Personal	4906	4906	Bedrock	12	Home
Wanda	Wenderly	Headquarters	Executives	Personal	4908	4908	Bedrock	14	Home
Wendy	Williams	Headquarters	Executives	Personal	4905	4905	Bedrock	11	Home
Yourst	Yumbal	Headquarters	Executives	Personal	4903	4903	Bedrock	9	Home

Figure 10-8 Individual Users List Page

If there are more users than can be displayed on one page, use the forward and backward arrows to page through the names and specify the number of records to display per page.

10.4.1 Individual Users List Page

The columns in the Individual Users list page are defined as follows:

- **First Name:** The first name of the user.
- **Last Name:** The last name of the user.
- **Site:** The site associated with the user.
- **User Group:** The user group that is associated with the user.

- **Client Type:** The Communicator type for which the user is authorized.
- **Extension:** The user's extension.
- **Mailbox:** The user's voice mailbox.
- **Switch:** The switch associated with the user.
- **Port:** The port associated with the user or the MAC address of an IP phone.
- **Status:** This shows the user's telephone port status.

  Home indicates that the user is at his or her home telephone port; Assigned indicates otherwise.

To add or edit a user's profile from the Edit User page, fill out the four tabs – General information, Personal Options for user options, Distribution Lists, and Workgroups – as described in the following sections. For information on importing user information from a spreadsheet, see the "User Import Tool" on page 350.

10.4.2 User Configuration Pages

10.4.2.1 General Page

General information about new and existing users is provided under the General tab on the Edit User page. Many data entry fields are automatically filled from other Director fields.

The following three subsections describe parameters on the General page.

Upper-Section Parameters

The parameters in the upper portion of the Edit User: General page, are shown in Figure 10-9.

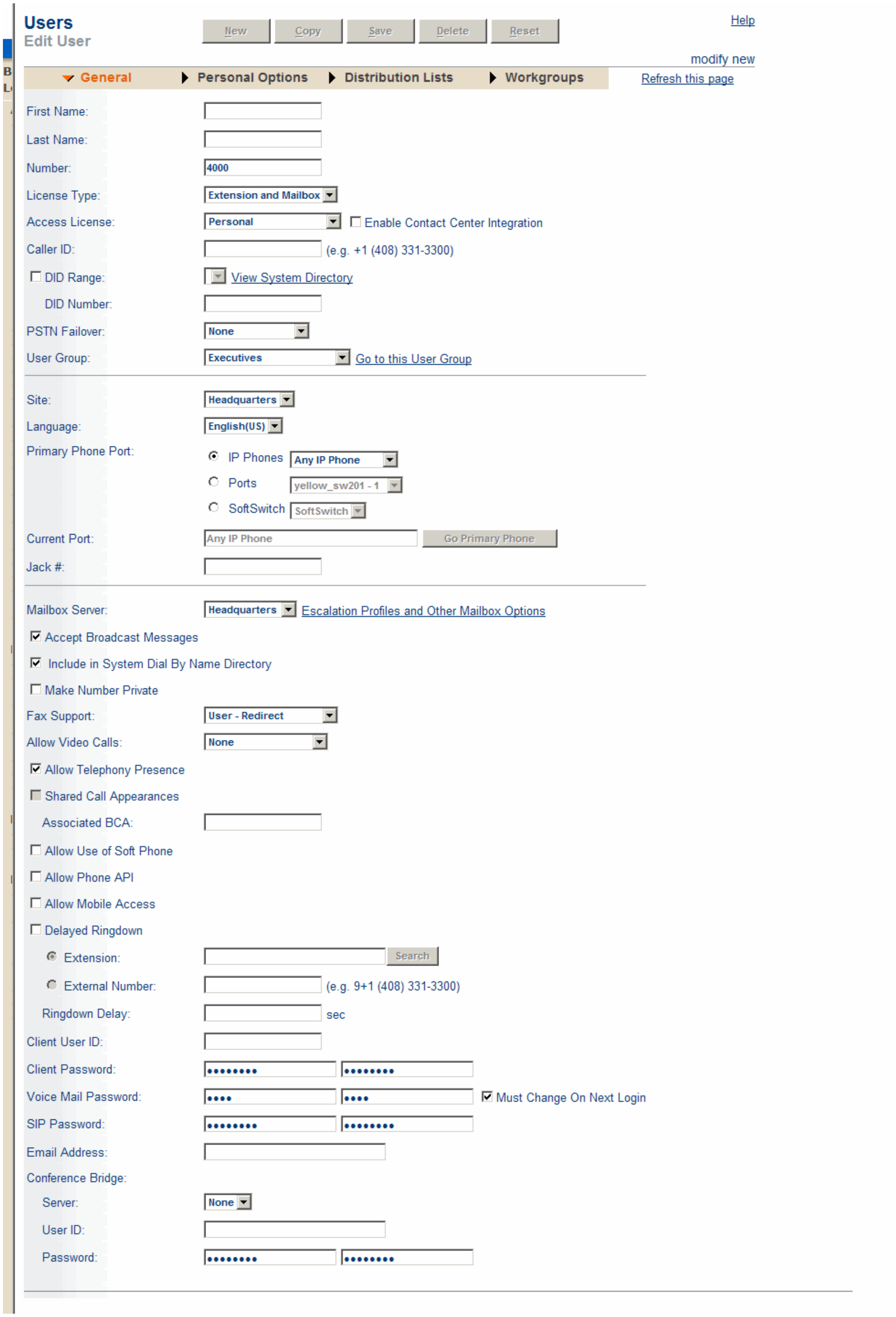


Figure 10-9 Edit User: General Page

The following are description of parameters located in the Edit User: General page:

- **First Name:** Specifies the first name of a user, fax machine, conference room, or "virtual user."
- **Last Name:** This is the last name of the user.
- **Number:** This is the user's extension.

  When configuring a new user, this field auto-fills with a number based upon a local cookie that was stored the last time a new user was configured.
- **License Type:** Set the user's license type from the following values: Extension and Mailbox, Extension-Only, and Mailbox-Only.
  - **Extension and Mailbox—**The user will have both a phone extension and an internal ShoreTel mailbox for voice mail.
  - **Extension-Only—**The user will have an extension but no mailbox. Use Extension-Only if, for example, the user has a mailbox for voice mail on a legacy switch or PBX. Selecting Extension-Only results in the following:
    - The voice mail server drop-down menu and associated check boxes are disabled.
    - User groups with SMDI ShoreTel voice mail are not available.
  - **Mailbox-Only—**The user will have a mailbox on the ShoreTel switch but will not have a phone extension. Selecting Mailbox-Only results in the following:
    - User groups with SMDI External voice mail are not available
    - Ports cannot be assigned to Mailbox-Only users.
    - Extension Assignment is not available to Mailbox-Only users, regardless of CoS settings.

    ShoreTel capacity is licensed by user license type. Make sure you have licenses for all users and required types.
- **Access License:** The access license field enables assignment of various levels of permissions for users using the ShoreTel Communicator application. Select from:
  - **Personal** (default)—This access level is available to all users and delivers desktop call control, visual voice mail, call history, and directory services, as well as options to control call handling and message notification. Since this is the default client configuration, it does not require a special license.
  - **Professional –** This access level provides access to all functions available through Personal Communicator plus Instant Messaging, Presence, Contact Viewer, and SoftPhone. A video license is also provided to users that have rights to Professional Communicator, providing access to VGA video.
  - **Workgroup Agent—**This access level is typically assigned to members of a workgroup and provides access to all features available through Professional Communicator, plus workgroup features—including login, logout, wrap-up, queue monitor, and shared workgroup mailbox—and the ability to transfer calls by dragging and dropping call cells into the buddy list. Does not include access to video.
  - **Workgroup Supervisor—**This access level is typically assigned to the supervisor of a workgroup and provides access to the agent monitor, in addition to all features available to Workgroup Agents.

  - **Operator—**This access level is typically assigned to operators, secretaries, and executive assistants. In addition to all features available in Workgroup Supervisor, it provides access to XGA Video and detailed information about destination extensions, including access to an extension monitor.
  - Enable Contact Center Integration

- **Caller ID**: This number is used for caller ID on outbound calls. A caller ID entered here will take precedence over the user's DID and the site's CESID number (both for normal outbound and 911 calls). If no number is entered, the user's DID or the site's CESID will be used for outbound caller ID.

  This feature is available only on outbound calls using a T1 PRI trunk.

- **DID Range:** This provides the user with a Direct Inward Dial (DID) number. This requires that DID services are properly configured against the desired trunk group. This is enabled by default if a DID trunk group is configured.

  - **DID Number:** In the field, enter the DID number that you want to assign to the user.

- **PSTN Failover:** Select from None, DID, or a specific phone number. When a specific phone number is to be entered, a text box opens. This field supplies the PSTN number to be dialed to complete a failed site-to-site call. When DID is selected and a call fails, the call is routed via the PSTN network to a local PSTN DID number for the user. When an external number is supplied and a call fails to go directly to its intended extension, the call is routed to the number indicated, for example, an auto-attendant.

  The user must have a Class of Service that has "Allow PSTN Failover" enabled. See the "Classes of Service" on page 314 for more information.

  If there is no available bandwidth or if a WAN is down for a site to site call and if the call destination has no PSTN failover, the call is directed to voice mail.

- **User Group:** This displays all user groups. Associate one group with the user. The system default is Executives. However, this default value may be modified using the **Preferences** menu of Director. Clicking Go to this User Group invokes the associated user group page.

- **Site**: This is the site for the user. This filters the list of switch ports that you must select from as well as provide a different default DID number if available.

- **Language:** Select the language this user will hear for voice mail prompts from the drop-down list of available languages.

- **Primary Phone Port:** This displays a list of available switch ports and IP phones. Select the switch port or IP phone to be associated with this user.

  If assigning an analog port and a selection is not specified, ShoreTel Director selects the next available port.

  Selecting IP Phone causes the drop-down list to display Any IP Phone as the default. For information about the Any IP Phone feature, see Chapter 12: Configuring User Features on page 409.

  If y ***Port*** is selected, the drop-down list displays ports available for phone use.

  To create a user without a port (a virtual user), select SoftSwitch as the home port.

  Assigning users to an analog port or SoftSwitch for their home port can cause the loss of phone service if the user selects the Go Home option in Communicator.

ShoreTel recommends that Extension Assignment users be assigned Any IP phone as their home port.

- **Current Port:** This field indicates the user's current switch port. This shows which switch port the user has assigned himself or herself. This field cannot be changed directly. Change the current port setting to the home port by clicking Go Home.

  Clicking Go Home causes the system to force the user back to his or her home telephone. This option is useful when a temporary user is no longer using that phone.

- **Jack #:** This is an informational field where the patch-panel jack number associated with the user's switch port may be recorded.

- **Mailbox on Server:** This enables the user's voice mailbox. The drop-down list allows selection of the server to host this mailbox. This is enabled by default.

  Changing a user's server requires the user exiting and re-launching the client to establish a connection to the new server.

  — **Escalation Profiles and Other Mailbox Options:**

- **Accept Broadcast Messages:** When enabled, this lets individual users receive broadcast messages. Disable this feature for users who do not want to receive broadcast messages. This is enabled by default.

- **Include in System Dial By Name Directory:** When enabled, this causes the user's name to be included in the auto-attendant's dial-by-name directory.

  This is enabled by default.

- **Make Number Private:** Checking this check box removes this number from the system directory and call handling destination lists.

- **Fax Support:** If a port is connected to a Fax machine and is in an environment where SIP PSTN gateways are used, this option can be configured such that the switch will freeze the jitter buffer and disable the echo canceller at the beginning of the call to ensure that FAXes are clearly and reliably transmitted. (Although jitter buffer and echo canceller enhance the quality of voice calls, they also impede reliable Fax transmission.) Options are:

  — ***User - No Redirect*** (Extension is connected to a user; do not redirect inbound Fax calls)

  — ***User - Redirect*** (Extension is connected to a user; redirect inbound Fax calls to site Fax redirect extension)

  — ***Fax Server*** (Extension is connected to a Fax server; do not redirect inbound fax calls but pulse DTMF digits)

  — ***Fax Machine*** (Extension is connected to a fax machine; do not redirect inbound fax calls)

  — **Non-T38 Data Terminal**

  — **Non-T38 Fax Server**

- **Allow Video Calls:** Selecting this option allows the user to perform video calls. Video calls is a user-licensed feature.

  — **None** Disables video capability for this user.

  — **Standard Resolution**

  — **High Resolution**

- **Allow Telephony Presence**: Telephony presence indicates a user's availability for accepting voice calls. Select this option to allow the user to access telephony presence information about other users.
- **Share Call Appearances:** This field allows a user to participate in Shared Call Appearances functions, assuming that a Bridged Call Appearance extension has been configured. See the"Bridged Call Appearance Conferencing" on page 250 for more information about Shared Call Appearances and Bridged Call Appearance configuration.
  - **Associated BCA**
- **Allow Use of SoftPhone:** Check this box to allow the user to have access to the SoftPhone option in Communicator.
- **Allow Phone API**: Third-party developers have the ability to create applications that can run on certain ShoreTel IP phone models. (Check with a ShoreTel representative for models that support this behavior.)

  Selecting this checkbox enables the IP phone associated with this user to go into PAPI browser mode, thus allowing the phone to run those third-party applications.
- **Allow Mobile Access:** Select this checkbox to enable Communicator for Mobile (CM) to run on this user's mobile device. The CM client software consists of a small Java applet that is installed on a mobile device, such as a BlackBerry. The CM software provides remote/mobile users with an interface similar to ShoreTel Communicator.

  Clear this checkbox to disable Communicator for Mobile on this user's mobile device.
- **Delayed Ringdown:** Select this checkbox to enable Delayed Ringdown for the extension. If this parameter is not enabled, all other Ringdown parameters are not selectable. Refer to Section 9.15.3.1 for information about Ringdown.
  - **Extension:** Select this option to specify a ShoreTel extension as the recipient device, then enter the extension of the recipient device in the corresponding data entry field.
  - **External Number:** Select this option to specify a phone outside of the ShoreTel network as the recipient device, then enter the phone number of the recipient device, including the Trunk Access Code of the ringdown trunk, in the corresponding data entry field.
  - **Ringdown Delay:** Enter the period of time that the phone waits after the user picks up the handset before initiating the ringdown call. Enter 0 in this field to cause the phone to immediately dial the ringdown device whenever the handset is picked u
- **Client User ID:** This is the login name that a user uses when logging into the ShoreTel Communicator or ShoreTel Director. By default, it is the first initial of the user's first name followed by the user's entire last name. This can be changed at your discretion.

  This field is automatically filled in when western language data is entered in the First Name and Last Name fields.
- **Client Password:** This is the password that a user will use when logging in to the system from the ShoreTel Communicator or ShoreTel Director. Characters that are entered in this field appear as asterisks.

The default password is `changeme`. Users are asked to change this password the first time they log in to the system.

It is recommended that this password NOT be changed because it is used by users who are configuring their Communicator for the first time.

- **Voice Mail Password:** This is the password that a user uses when logging in to his or her voice mailbox from the telephone. Characters that are entered in this field appear as asterisks.

  The default password is `1234`. Users are asked to change this password the first time they log into the system.

  It is recommended that this password not be changed because it is used by users who are configuring their telephone for the first time.

  — **Must Change on Next Login:** This check box is selected by default when a new user is created. This forces the user to enter a new password the first time they log into their mailbox. Once the user has entered a new password, the system clears this checkbox.

  If a user has forgotten his or her password, the system administrator can reset this option (i.e. select the checkbox) and enter a generic password, thus allowing the user to re-enter a new password.

- **SIP Password**: Entering a value in the data entry box enables the extension to support SIP. Clearing the SIP Password data fields disables the extension from supporting SIP. Refer to Section 18.2.6.5 for more information.

- **Email Address**: This is the user's e-mail address. By default, it is automatically entered when you enter the user's first and last names in the First Name and Last Name fields. It consists of the first initial of the user's first name followed by the user's entire last name—for instance, `sdemont`. In addition, the `@companyname.com` domain is saved in a cookie on your workstation each time you save a user. This information is then presented as a default, which can be changed at your discretion.

  Be sure to delete this field if the user does not have or use e-mail.

- **Conference Bridge:** This section allows you to assign the user to an Service Appliance 100 (SA-100) conference device.

  — **Appliance:** In the field, select the SA-100 that you want the user to use.

10.4.2.2 Personal Options

To configure new and existing users' personal options, click the Personal Options tab (shown in Figure 10-10) on the Edit User page. After making entries, click the link at the bottom of the page to return to the user's edit page.

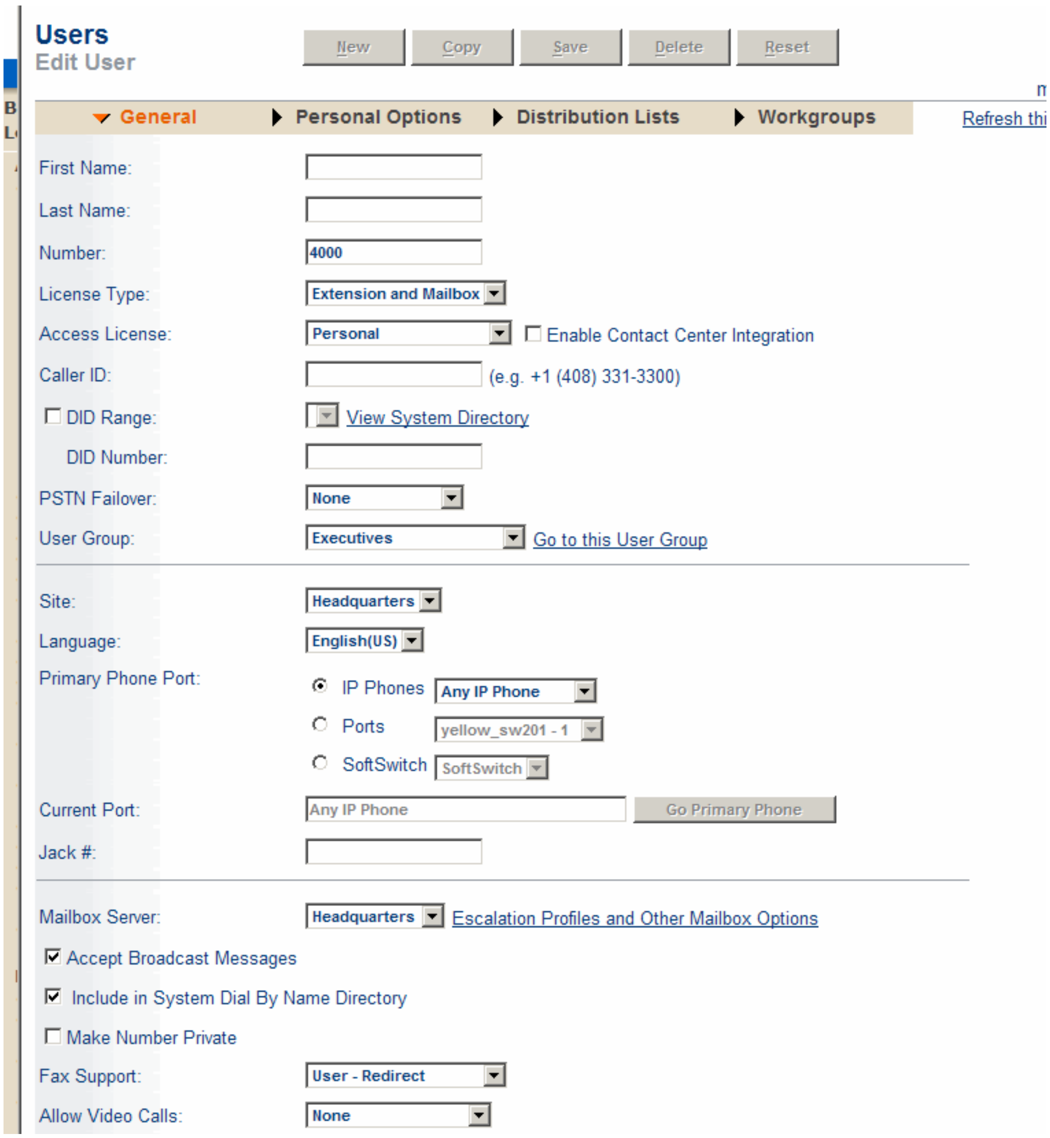


Figure 10-10 Edit User Page – Personal Options

The parameters under the Personal Options tab are as follows:

CALL CONTROL OPTIONS

- **Current call stack size:** A call stack is the ShoreTel extension component that manages an entity's call appearances. The call stack size defines the maximum number of calls – including active and held calls – that an extension can handle simultaneously. When this number is exceeded, calls are either given a busy tone or forwarded, depending on the call handling mode that is currently in effect.

  Valid values are 1 through 16 with the limitation that the value specified in the call stack size of the user's CoS becomes the upper limit.

- **Ring Type:** Select from the drop-down list of one of the four different ring types. The ring type applies to ShoreTel IP Phones but not to analog telephones.

NOTE Ring type is a personal option for the user and not a configuration setting of the phone. When a user moves from phone to phone, their ring type will follow them.

Also, keep in mind that most ShoreTel IP phone models support the ability to load customized ring tones onto each phone, so that each user can have a unique ring tone, if desired. See the "Custom Ring Tones" on page 205 for details.

- **Wall Paper:** This field allows the user to select the “wallpaper” or background appearance of their Communicator program.
- **Automatic Off-Hook Preference:** This option is intended for users who rely primarily on a headset (standard or wireless). This feature eliminates the requirement to manually press the Headset button on the IP phone to activate the headset.
  - **Speakerphone:** Automatically route incoming or outgoing calls to the speakerphone. This is the default.
  - **Headset:** Automatically route incoming or outgoing calls to the headset.
  - **Wireless Headset:** Automatically route incoming or outgoing calls to the wireless headset.
  - **Bluetooth Headset:** Automatically route incoming or outgoing calls to the Blootooth headset.

  For configuration instructions, refer to "Wireless Headset Hook Switch" on page 210.
- **Handsfree Mode:** This enables the user’s handsfree mode. When enabled, dial tone is disabled so that the user can use a headset or speakerphone to answer and make calls from the desktop client.

  Handsfree mode is useful for both headset and speakerphone users.

  The default is disabled.
- **Call Waiting Tone Enabled:** This enables the user’s call waiting tone. When enabled, the user hears the tone from the telephone handset when a second call is waiting.

  The default is enabled.

  Subtle differences can be heard in the call waiting tone one hears for a direct extension as opposed to a monitored BCA extension. This difference has been implemented to help differentiate between the two.
- **Trunk Group Access Code:** This lets you assign the default trunk access code for this user. Although this actually lists trunk groups, the only parameter leveraged is the access code. If the user has access to multiple trunk groups with the same access code, these trunks are seized based upon the Network Call Routing feature operation described in the ***ShoreTel 12: Planning and Installation Guide***.

  This eliminates forcing every user to configure his or her desktop client with the proper access code for dialing.
- **Mailbox for Recorded Calls:** Call can be monitored or recorded by people with appropriate permissions. Indicate the mailbox to be used for recordings. The maximum length of recording is determined by the voice mail class of service for the destination mailbox.

- **Program IP Phone Buttons:** This link takes you to the Program IP Phone Buttons page where you can assign functions to the ShoreTel IP Phone or BB24 programmable buttons.

  For configuration instructions, refer to "Programmable IP Phone Buttons" on page 211.

- **Copy:** This button takes you to the Copy IP Phone Buttons page. The Copy button reduces the tedious work of configuring the programmable buttons on each user's IP phone by allowing a system administrator to copy a programmable button configuration from one user to another.

  For configuration instructions, refer to the "Copying Programmable Buttons Configurations" on page 215.

- **Program Communicator Toolbars:** This link takes you to the Program Communicator Toolbars page where you can create new toolbars and assign common functions to the buttons on that toolbar.

  The customizable toolbars that extend across the top of the Communicator user interface. The buttons in these toolbars can be programmed with common operations in a manner similar to the custom buttons on some IP phone models. Communicator users can then perform many operations (e.g. "speed dial" or open URL) just by clicking a button on the toolbar.

  For configuration instructions, refer to the "Creating a Personal Programmable Toolbar" on page 375.

- **External Assignment:** This link takes you to the Find Me and External Assignment page where you can set two Find Me numbers as forwarding destinations, or where you can enter an external PSTN phone number for use with the Extension Assignment feature.

  For configuration instructions, see the "Find Me and External Assignment Page" on page 419, or just refer to Chapter 12 for related topics.

- **Personalized Call Handling Rules:**

CALL HANDLING MODE OPTIONS

- **Current Call Handling Mode:** This lets the system administrator set the current call handling mode for the user. Subsequently, users can select and edit each call handling mode from their desktop client.

  The default is Standard.

  For more information about call handling, see the "Automated Call Handling" on page 421.

- **Delegation:** You have the option of delegating the call handling management to one or more other users. This feature is helpful for users who may want their personal assistants to change their call handling mode for them.

- **Outlook Automated Call Handling:** This lets a user's Microsoft Outlook Calendar control his or her call handling mode. Select this check box to enable this feature.

- **Edit Call Handling Modes:** The links under this section bring you to the call handling mode configuration pages. Call handling modes specify how, when, and where calls are forwarded, and whether the user requires message notification when voice mail is received.

MAILBOX OPTIONS

- **Find Me:** This link takes you to the Find Me and External Assignment page where you can set two Find Me numbers as forwarding destinations or where you can enter an external PSTN phone number for use with the Extension Assignment feature.
- **Escalation Profiles and Other Mailbox Options:** This link takes you to the Escalation Profiles and Other Mailbox Options page where you can create Message Notification Escalation Profiles, configure Email Delivery Options for a user, and set up Automatic Message Forwarding.

  For configuration instructions, see the "Escalation Profiles and Other Mailbox Options" on page 448.

10.4.2.3 Distribution Lists

A distribution list lets a user easily send a voice mail message to multiple users at one time. Each distribution list has a descriptive name and distribution list number associated with it.

See the "System Distribution Lists" on page 441 for details about adding and populating Distribution lists.

Users can be associated with the distribution lists from either the Distribution Lists options on the Edit User page (Figure 10-11) or the System Distribution List edit page. When a user is associated with a distribution list, that user receives messages sent to that list.

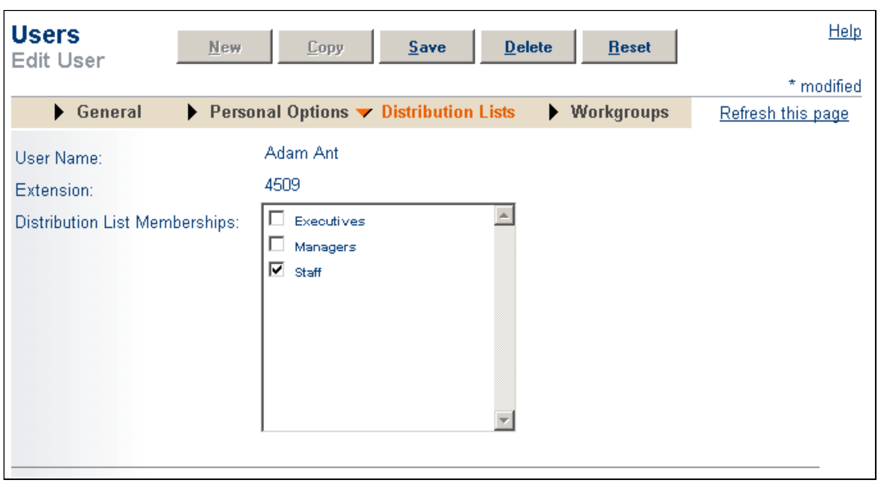


Figure 10-11 Edit User Page (Distribution Lists Tab)

If the user clicks the Distribution Lists tab on the Edit User page, the System Distribution List page appears.

The Distribution List Memberships box shows the distribution lists that are currently available. Clicking their check boxes adds or removes the user from the indicated list.

Users can be associated with more than one distribution list.

10.4.2.4 Workgroups

The Workgroups tab on the Edit User page allows editing a user's workgroup membership, as shown in Figure 10-12. Users can belong to multiple workgroups; however, a user's login status is the same for all workgroups of which he is a member.

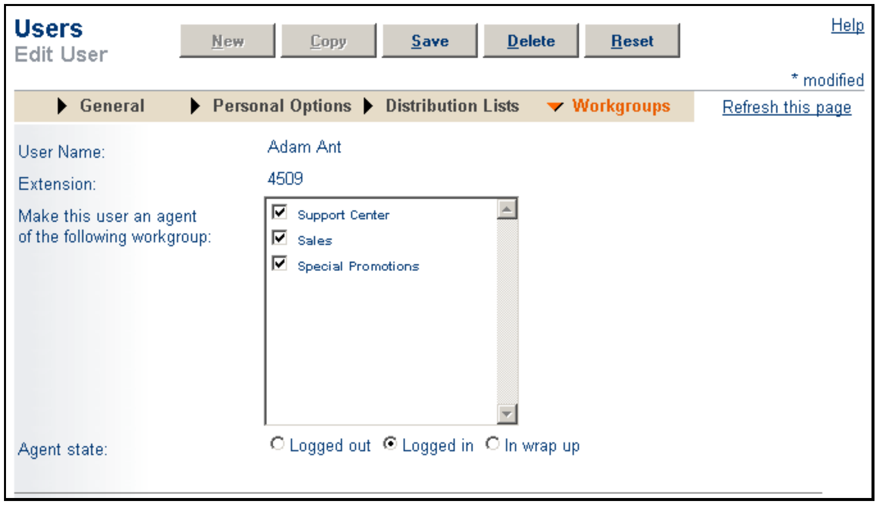


Figure 10-12 Edit User Page (Workgroups Tab)

The Workgroups box shows the workgroup lists that are currently available. Clicking the check boxes adds or removes the user from the indicated list. Click the Logged In check box to activate membership.

10.4.3 System Directory Record

When adding a new user to the system, a user entry is automatically made to the system directory. Edit a user's system directory record by going to the System Directory Entry edit page, shown in Figure 10-13.

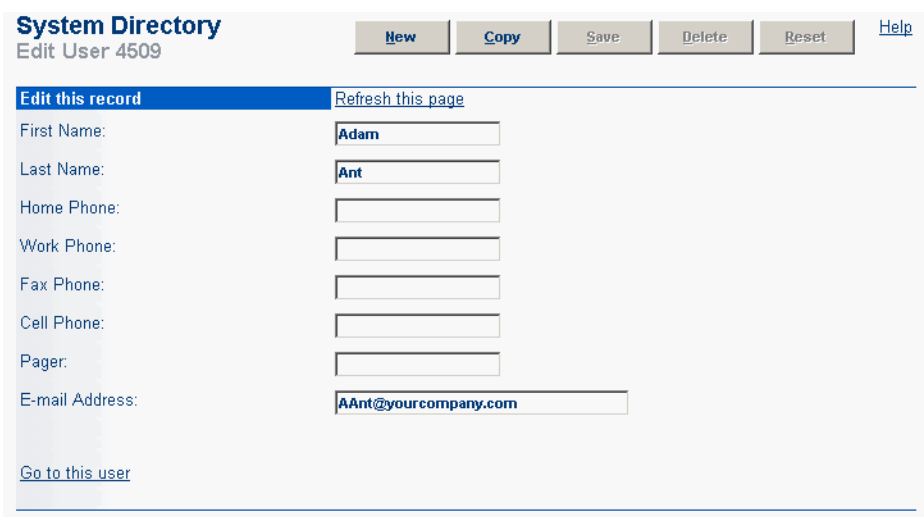


Figure 10-13 System Directory Edit Entry Page

Edit some or all of the fields on the System Directory Entry edit page as needed. The First Name, Last Name, and E-mail Address fields are automatically entered when adding a new user. Fields modified on either page are automatically updated on the other page.

10.5 Active Directory

Directory services store organization information and settings in a central, organized, accessible database. Active Directory (AD) is the Microsoft application that implements AD on Windows based systems. AD is widely deployed among large enterprises.

The ShoreTel AD implementation supports the synchronization of user records between the ShoreTel database and other applications using Active Directory on Windows based networks.

ShoreTel AD includes the following features:

- Authentication of AD Users, as described in "Authenticating AD Users" on page 345.
- Synchronizing AD Directory and ShoreTel Director user records, as described in "Synchronizing AD and Director User Records" on page 347.
- Bulk Provisioning of User Accounts, as described in "Bulk Provisioning of User Accounts" on page 348.

10.5.1 Active Directory Implementation

Active Directory Integration is an optional ShoreTel feature that is disabled by default. Systems that disable AD Integration do not recognize links to the Active Directory for properties attached to system users and do not provide AD authentication, synchronization, or provisioning services.

When AD Integration is enabled, only users that have administrative permissions can log into ShoreTel Director. This requires the explicit definition of at least one administrator role and the assignment of users requiring ShoreTel Director access to an administrative role.

10.5.1.1 Enabling AD Integration on a ShoreTel System

Active Directory is enabled from the Edit Other System Parameters page, as shown in Figure 10-14, is accessed through ShoreTel Director by selecting Administration -> System Parameters -> Other. To enable Active Directory, select Enable AD Integration located at the bottom of the page, then enter the Active Directory path in the data entry field.

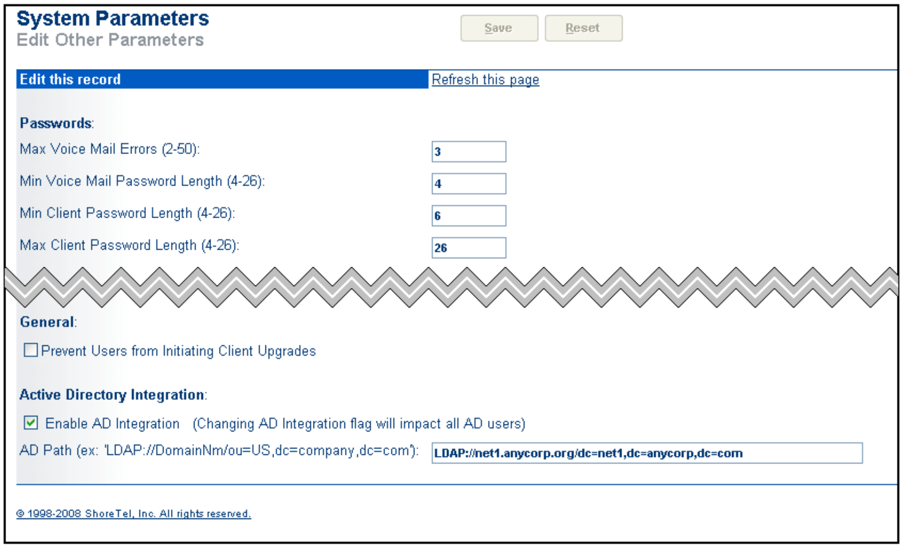


Figure 10-14 Enabling Active Directory

IMPORTANT Since AD prevents anonymous users from logging in through the default admin account, at least one user must be assigned Administrative rights before enabling AD.

"Configuring AD on a New System" on page 343 provides a procedure that includes required user and administrative assignment steps for enabling Active Directory on a newly installed ShoreTel system.

10.5.1.2 Configuring AD on a New System

The following procedure enables AD Integration on a new system, includes initial user configuration and assignment of administrative permissions:

Step 1 Login to an Active Directory account through which ShoreTel Director will be accessed.

Step 2 Access ShoreTel Director through the following parameters:

username – admin
password – changeme

Step 3 Add a user to the ShoreTel system with Client User ID name that matches the Active Directory login name.

To add a system user, select **Administration -> Users -> Individual Users** from the Main Menu. Refer to the ShoreTel Administration Guide for more information.

Step 4 Assign Administrative Rights to the new user account.

To assign rights, press the New button in the Administrators List page that is accessed by selecting Administration -> System Parameters -> Administrative Permissions -> Administrators page.

Step 5 Open the System Parameters: Edit Other Parameters page by selecting **Administration > System Parameters > Other** from the Director Main Menu Login.

Step 6 Click **Enable AD Integration** located at the bottom of the page, then enter a valid AD path in the **AD Path** data entry field.

Step 7 When prompted to save your changes and close the window, press the OK button and yes option.

This step logs you out of Director

Step 8 Log into Director again, using the username created in Step 3.

The default password is changeme.

Step 9 Open the Edit User page for the new user and enable Active Directory for that user, including the domain and user id for the user.

Step 10 Log out of ShoreTel Director and close the window.

Subsequent ShoreTel Director launches when logged on to the domain through this username will automatically log into Director.

10.5.1.3 Mapping Active Director Attributes to ShoreTel Fields

ShoreTel user records contain eleven data fields that map directly to Active Directory records. The following is a list of these data fields, categorized by the Director page that sets their ShoreTel value

Edit User Page

- **First Name**: Active Directory field capacity is 64 characters; ShoreTel capacity is 50.
- **Last Name**: Active Directory field capacity is 64 characters; ShoreTel capacity is 50.
- **Number**: ShoreTel extracts the number of digits specified by the extension length, starting from the right side of the AD phone number.
- **DID**
- **Email Address**
- **Client User ID:** Active Directory length is limited to 20 characters.

System Directory

- **Home Phone**
- **Pager**
- **Cell Phone**
- **Fax Phone**

ShoreTel Database (does not appear in Director)

- **LDAP-GUID**: Used internally by the ShoreTel system when performing subsequent user updates from the AD database.

10.5.1.4 Authenticating AD Users

ShoreTel supports AD authentication for users logging into Communicator, Web Client, and Director, permitting access to these programs without providing the ShoreTel username or password.

- AD Users logging into Director and Communicator are authenticated through Single Sign On (SSO) with their current network credentials. Users are not required to re-enter their credentials to access these applications.
- AD users logging into Web Client are authenticated through Explicit Authentication, which requires re-entry of their credentials each time they access the application.

10.5.1.5 Director

When AD Integration is enabled, user access to ShoreTel Director is restricted as follows:

- Only users with a domain account can log into Director
- Only users with administrative permissions can log into Director
- Users do not need to be logged into their domain account to access Director
- Users do not need their ShoreTel account configured for Active Directory (AD Users) to access Director

The following sections describe Director access scenarios.

AD User Logged into the Domain

Users configured for AD within ShoreTel that are logged into the domain network are directed to the Director Quick Look page when they attempt to access Director without entering their network credentials.

Upon logging off from Director, the browser displays a ShoreTel entry page that allows Director access by pressing a button, as shown in Figure 10-15.

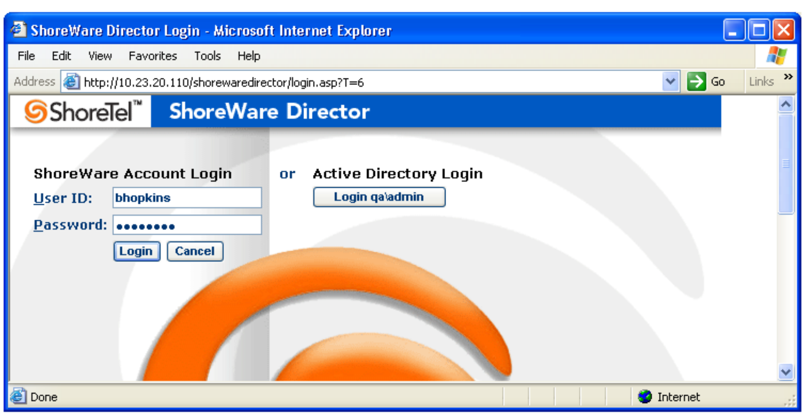


Figure 10-15 Director Login Page – AD user

Non-AD User Logged into the domain

Users that are not configured for AD within ShoreTel that are logged into the domain network are directed to a Director Login page, as shown in Figure 10-16. Users access Director from this page by entering their ShoreTel username and password in the respective data fields.

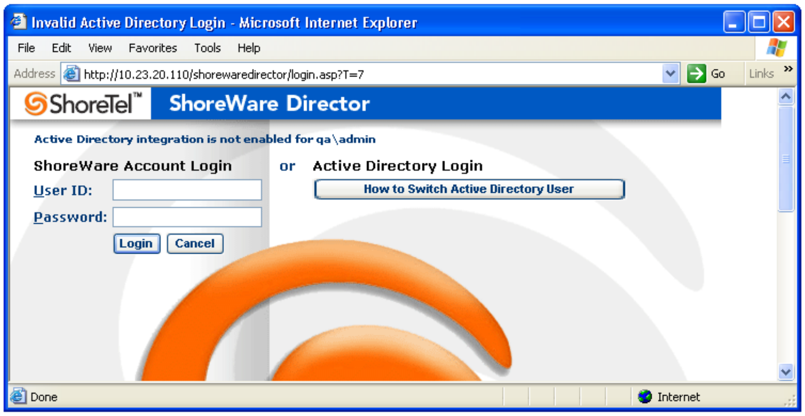


Figure 10-16 Director Logon Page – Non-AD user

Users not logged into the domain

Users attempting to access Director without first logging into the domain network are initially routed to a domain login page, as shown in Figure 10-17. After entering their network credentials, users are routed to the appropriate page – AD users are routed to the Quick Look page and Non-AD users are directed to a login page.

Figure 10-17 Domain Login Page

10.5.1.6 Communicator

When AD Integration is enabled, access to Communicator is available all system users, including users without domain accounts or not configured as ShoreTel AD users.

Initial Configuration

When installing Communicator, users authenticated through AD are not queried for their credentials; they are immediately prompted for the server name after which wizard panels guide them through the setup process.

Users that are not authenticated through AD are required to enter their name, password and server name. After verifying the user's credentials, Communicator guides the user through the setup process.

Logging into Communicator

Attempts to log into Communicator after the initial setup are handled on the basis of the user's AD configuration. Active Directory users are authenticated by SSO through the verification of their AD credentials. Users not configured for Active Directory are authenticated through the verification of their ShoreTel username and password, as previously loaded through their Communicator account. In either case, users are typically not required to re-enter their username or password each time they open Communicator from their computer.

Communicator behavior after an authentication failure is not changed by this feature.

User Name – Telephony Options Page

The User Name data field at the top of the Options and Preferences: Telephony page displays the User name of the Communicator account. This field is read only for users authenticated through AD credentials, as shown in Figure 10-18.

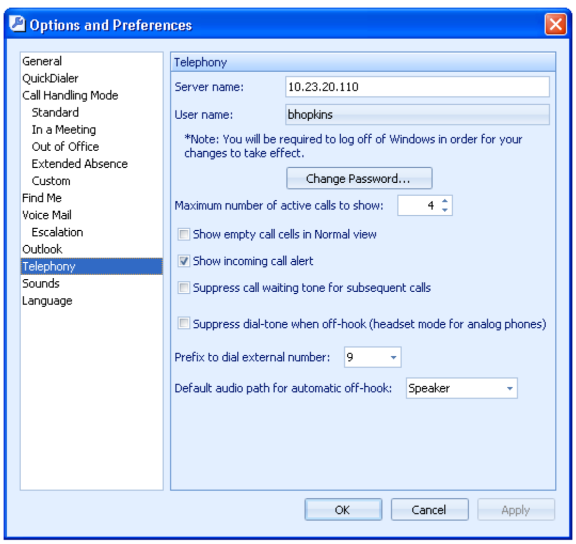


Figure 10-18 Options and Preferences: Telephony Page

10.5.1.7 Web Client

When AD Integration is enabled, access to the Web Client is available all users systems, including users without domain accounts or not configured as ShoreTel AD users.

Attempts to log into Web Client always require the user to enter credential information. Domain members configured as ShoreTel Active Directory members enter their Active Directory user name and password and are authorized through Explicit Authentication. All other users enter their ShoreTel username and password.

10.5.2 Synchronizing AD and Director User Records

Director provides an interface for adding, updating, and deleting AD users from the ShoreTel database. Synchronization is performed on individual users and does not affect the AD directory.

10.5.2.1 Enabling AD for a ShoreTel User

When AD Integration is enabled, the first parameter the Edit User: General page enables Active Director for the user.

To enable Active Director for a user, enable the Active Directory User option at the top of the page, then enter the domain\user name for the user, as shown in Figure 10-19.

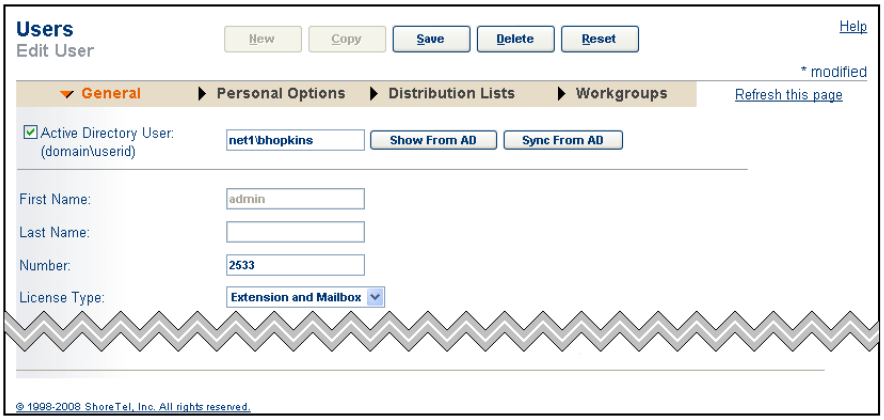


Figure 10-19 Enabling Active Directory for a User

10.5.2.2 Updating AD Fields

Active Directory users can synchronize user account records with contents from the Active Directory database by pressing the Sync from AD button located left of the user's Active Directory userid. Pressing the Show From AD button displays parameter settings for the users; Active Directory account.

The Show From AD and Sync From AD buttons are inactive for users accessing this page that are not configured as AD users

10.5.2.3 Removing AD Users

When an administrator attempts to delete a user with an AD account, ShoreTel displays a warning message and requires confirmation before removing the record from the ShoreTel database.

10.5.3 Bulk Provisioning of User Accounts

ShoreTel supports the bulk provisioning of user accounts from AD records through a two step process:

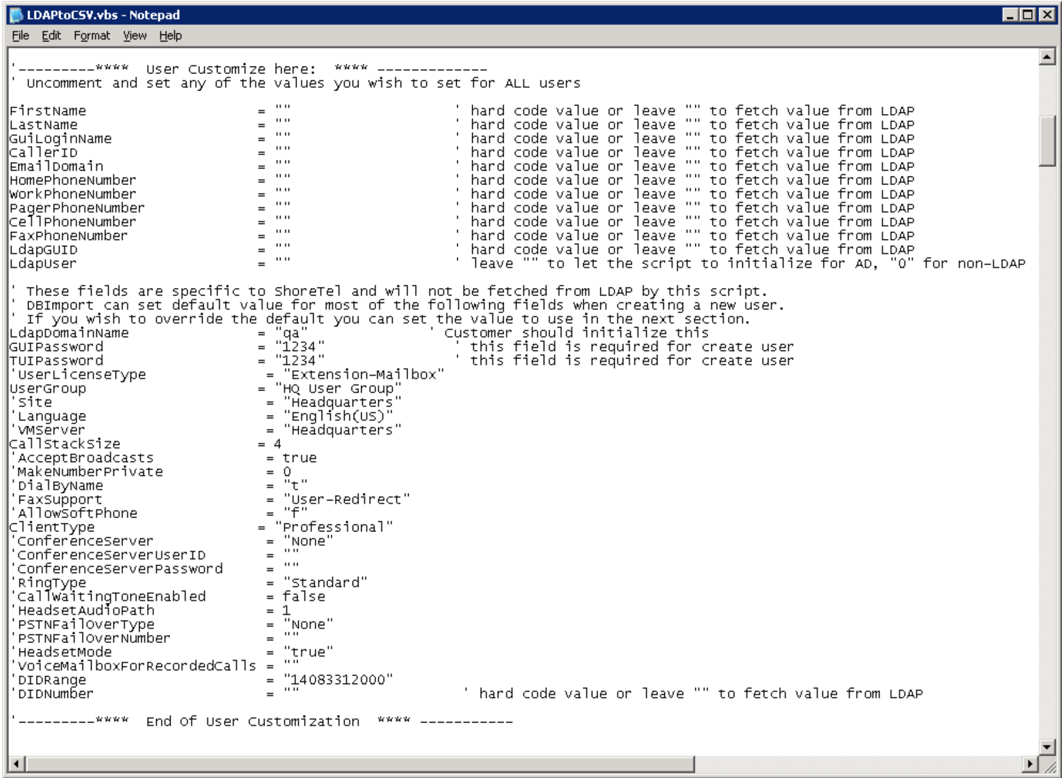
Step 1 Export of AD records to a CSV file.

Step 2 Import CSV records to the ShoreTel user database.

10.5.3.1 Exporting AD Records

ShoreTel provides a VBScript file to export data records from an Active Directory database to a CSV file. The VBScript file can be used as a template for specific ShoreTel system requirements.

The parameter section of the file is shown in Figure 10-20. This section consist of two subsections:


LDAPtoCSV.vbs - Notepad

File Edit Format View Help

```
'---------****  User Customize here:  **** -------------
' Uncomment and set any of the values you wish to set for ALL users

FirstName                    = ""                    ' hard code value or leave "" to fetch value from LDAP
LastName                     = ""                    ' hard code value or leave "" to fetch value from LDAP
GuiLoginName                 = ""                    ' hard code value or leave "" to fetch value from LDAP
CallerID                     = ""                    ' hard code value or leave "" to fetch value from LDAP
EmailDomain                  = ""                    ' hard code value or leave "" to fetch value from LDAP
HomePhoneNumber              = ""                    ' hard code value or leave "" to fetch value from LDAP
WorkPhoneNumber              = ""                    ' hard code value or leave "" to fetch value from LDAP
PagerPhoneNumber             = ""                    ' hard code value or leave "" to fetch value from LDAP
CellPhoneNumber              = ""                    ' hard code value or leave "" to fetch value from LDAP
FaxPhoneNumber               = ""                    ' hard code value or leave "" to fetch value from LDAP
LdapGUID                     = ""                    ' hard code value or leave "" to fetch value from LDAP
LdapUser                     = ""                    ' leave "" to let the script to initialize for AD, "0" for non-LDAP

' These fields are specific to ShoreTel and will not be fetched from LDAP by this script.
' DBImport can set default value for most of the following fields when creating a new user.
' If you wish to override the default you can set the value to use in the next section.
LdapDomainName               = "qa"               ' Customer should initialize this
GUIPassword                  = "1234"                ' this field is required for create user
TUIPassword                  = "1234"                ' this field is required for create user
'UserLicenseType              = "Extension-Mailbox"
UserGroup                    = "HQ User Group"
'Site                         = "Headquarters"
'Language                     = "English(US)"
'VMServer                     = "Headquarters"
CallStackSize                = 4
'AcceptBroadcasts             = true
'MakeNumberPrivate            = 0
'DialByName                   = "t"
'FaxSupport                   = "User-Redirect"
'AllowSoftPhone               = "f"
ClientType                   = "Professional"
'ConferenceServer             = "None"
'ConferenceServerUserID       = ""
'ConferenceServerPassword     = ""
'RingType                     = "Standard"
'CallWaitingToneEnabled       = false
'HeadsetAudioPath             = 1
'PSTNFailoverType             = "None"
'PSTNFailoverNumber           = ""
'HeadsetMode                  = "true"
'VoiceMailboxForRecordedCalls = ""
'DIDRange                     = "14083312000"
'DIDNumber                    = ""                     ' hard code value or leave "" to fetch value from LDAP

'---------****  End Of User Customization  **** -----------
```

Figure 10-20 Sample ldaptocsv.vbs file – parameter sections

- The Top Section receives values from the AD database for each imported file. As indicated in the comments section, one value can be assigned to individual fields for all users by entering the value in place of the quotes.
- The Bottom section is a data template containing non-AD fields that are saved to the CSV file. The values entered in these fields are assigned to all records retrieved from the AD database.

The following command line executes the AD to CSV file import:

cscript ldaptocsv.vbs *outputCSVfile LDAPpath [modifiedInLastN#OfDays]*

where

outputCSVfile specifies the name of the output file
LDAPpath specifies the path to the AD database
[modifiedInLastN#OfDays]

The resulting CSV file can be modified though a spreadsheet program to customize user settings for import to the ShoreTel database.

10.5.3.2 Exporting the CSV File

DBImport.exe is a executable file that imports CSV contents to the ShoreTel database that was introduced in a previous ShoreTel release and described in the ShoreTel Administrator Guide.

DBImport supports SIP servers. SIP server names can be entered in the pre-existing voicemail server field. DB Import determines the user licence type and user group voicemail interface mode to use for looking up the voicemail server name.

To support AD record imports, the following fields are supported by DBImport.exe:

- **LDAP-User flag**: a indicator that the user settings were imported from an AD database.
- **LDAP-GUID**: a data field that creates an association between an AD record and the corresponding ShoreTel record.
- **LDAP-DomainName**: specifies the name of the user's domain network

The following command line imports CSV file records to the ShoreTel database.

Dbimport -log DbLog.log -err DbErr.err <inputCSVfile>

10.6 User Management Utilities

10.6.1 User Import Tool

The database import utility allows a system administrator to make changes across all users in a system with a single action. This new feature improves the ease with which a system administrator can modify information for large groups of users.

The system administrator creates a spreadsheet using a common application, such as Microsoft Excel. He or she modifies the user data in the spreadsheet in "free form" fashion, instead of having to modify each user's information, one at a time, from within Director. For someone who is skilled at using spreadsheet software, the time savings can be enormous.

Once the user information has been manipulated within the spreadsheet, it can be exported to a CSV format and this CSV file can then be imported into the ShoreTel system.

Details:

- The import tool supports modify/delete/insert operations, allowing a system administrator to add users, delete users, or modify the account of an existing user.
- Using the import tool does not require scheduled downtime. However, when importing large files that contain many rows of information, performance may be affected as the database is frequently queried. So depending on the size of the imported file and the type of information that is being added or modified, it may be recommended to perform the import during off hours.
- The ShoreTel system does not currently support the ability to export user data into a CSV file.
- The Import utility does not allow the administrator to specify the Users' Home Port. The Home Port parameter is automatically set to SoftSwitch within each User record added or modified by an import.

Table 10-1 lists and provides information about field headers that must appear as the first row in the CSV file that will be imported. The field order can be random, but all fields are mandatory, case sensitive, and must have data.

Table 10-1 Field Headers for CSV File

DataBase Import Field	DataBase Field Name	Accepted Input
Extension	UserDN	
FirstName (String)	TabAddresses.FirstName	
LastName (String)	TabAddresses.LastName	
GuiLoginName (String)	GuiLoginName	
GUIPassword (String)	GUIPassword	
UserLicenseType (Code)	LicenseTypeID	Extension-Mailbox Extension-Only Mailbox-Only
CallerID (String)	CIDNumber	A full canonical number (such as +1 (408) 331-3300)
UserGroup	UserGroupID	(Code. Must be one of the existing User groups configured on the system.)
Site	Site	(String Name - Must be one of the existing Site Name)
Language	DN.LanguageID	Value should be a language name (e.g. "English(US)").
VMServer	VMServerID	Value should be a VM server, SIP server, or QSIG server name, depending on user's user group interface mode.
CallStackSize	CurrentCallStackDepth	Number
AcceptBroadcasts	Mailboxes.NoReceiveBroadcasts	Boolean DB value is opposite of input value
MakeNumberPrivate	DN.Hidden	Boolean
DialByName	DN.ExcludeFromDialByName	Boolean DB value is opposite of input value
Fax Support	FaxSupport	"User-No Redirect" "User-Redirect" "FAX Server" "Fax Machine"
AllowSoftPhone	AllowSoftPhone	Boolean

Table 10-1 Field Headers for CSV File

DataBase Import Field	DataBase Field Name	Accepted Input
ClientType	ClientType	Personal Professional Workgroup Agent Workgroup Supervisor Operator Contact Center Supervisor Contact Center Agent
EmailDomain	TabAddresses.EmailAddress	String
ConferenceServer	BridgeID	String Name - Must be the existing Conference Bridge Name or “None”
ConferenceServerUserID (String - User ID for the conference bridge)	BridgeUserID	
ConferenceServerPassword (String)	BridgePassword	
RingType	RingToneID	Value must be a ring tone name: Standard Ring 2 Ring 3 Ring 4
CallWaitingToneEnabled	CallWaitingToneEnabled	Boolean
HeadsetAudioPath	UseHeadsetAudioPath	Number
HeadsetMode	HeadsetMode	Boolean
PSTNFailOverType	PSTNFailOverTypeID	None External DID
PSTNFailOverNumber	PSTNFailOverNumber	A full canonical number (such as +1 (408) 331-3300)
DIDRange	DIDDigitMap.DIDRangeID	A full canonical number base of range (such as +1 (408) 331-3300)
DIDNumber	DIDDigitMap.Digits	A full canonical number (such as +1 (408) 331-3300)
VoiceMailboxForRecordedCalls	VoiceMailboxForRecordedCalls	An existing valid extension (or can be left blank)

Table 10-1 Field Headers for CSV File

DataBase Import Field	DataBase Field Name	Accepted Input
MustChangeTUIPassword	MustChangeTUIPassword	Boolean
MustChangeGUIPassword	MustChangeGUIPassword	Boolean
EnableCC	ContactCenterIntegration	Boolean
MustRecordName	MustRecordName	Boolean
HomePhoneMACAddress		
HomePhoneNumber		A full canonical number base of range (such as +1 (408) 331-3300)
WorkPhoneNumber	TabAddresses.WorkPhone	A full canonical number base of range (such as +1 (408) 331-3300)
PagerPhoneNumber	TabAddresses.PagerPhone	A full canonical number base of range (such as +1 (408) 331-3300)
CellPhoneNumber	TabAddresses.CellPhone	A full canonical number base of range (such as +1 (408) 331-3300)
FAXPhoneNumber	TabAddresses.FAXPhone	A full canonical number base of range (such as +1 (408) 331-3300)
LdapUser	LDAPUser	“Non-LDAP user” “Active Directory”
LdapGuid	LDAPGuid	String
LdapDomainName	LDAPDomainName	String
AllowPapi	AllowPAPI	Boolean
MCMAllowed	MCMAllowed	Boolean
MCMPinEntryFrequency	MCMPinEntryFrequency	Number
AllowVideoCalls	AllowVideoCalls	Boolean
AllowTelephonyPresence	AllowTelephonyPresence	Boolean
IMServer	IMServerID	Value should be an IM server name. “<NULL>” clears the field.

Additional considerations:

- String names fields must exist in the system. For example, if a new user is to be created in site New York. “New York” must exist as a site name. String names are case sensitive.
- For fields that are code in nature, specify the description to be displayed as it appears in Director. Data validation will translate the displayable value to the appropriate code. Note that the descriptions are case sensitive.

- Boolean fields can be true/false or 1/0.
- All fields are mandatory when adding new users.
- When updating an existing user, fields left blank will not change existing values.

Example: Table 10-21 shows a sample of the spreadsheet values before the Director import.

1	Extension	FirstName	LastName	GuiLoginN	GUIPassw	TUIPassw	UserLicens	CallerID	UserGroup	Site	Language	VMServer	CallStackS	AcceptBro
2	233	Jane	Doe	Changeme	1234	3456	Mailbox-Only		Executives	Headquart	English(US	Headquart	2	TRUE

Figure 10-21 Spreadsheet values before Director import

The following procedure describes the process of importing data from a CSV file into the user database. This procedure assumes that you have already successfully exported the data from the spreadsheet into a CSV file, and stored that file in the proper location.

Step 1 Verify that the CSV file to be imported is located on the HQ server and in the following location:
C:\Program Files\Shoreline Communications \ ShoreWare Server

Adding Users

- These fields are required to add a user with DB Import: FirstName, LastName, GUIPassword, TUIPassword, GUILoginName.
- The Extension field is optional when adding a user. If not provided, it will be automatically assigned.
- Any blank or omitted fields will be left unchanged.

Updating Users

- “Enter the Extension and any fields to be updated.
- “Any blank or omitted fields will be left unchanged.

Deleting Users

- “Enter the Extension and leave all other fields blank to delete a user.

Step 2 Open the command prompt window in the directory shown above and run the following command:

```
Dbimport -log DbLog.log -err DbErr.err <CSV-file>
```

- -err is the flag to create an error log file.
- error.log is the file where error messages will be stored.
- CSV-file is the name of the file.

The import adds any new users and modifies any existing users in the user database as indicated in the CSV file.

When a user is updated, he or she is assigned to “Any IP Phone” on the headquarters server.

10.6.2 Batch Update Utility

The Batch Update Utility lets you make changes to multiple users at the same time. It allows you to find a set of users and globally change certain parameters. Due to the scope of this change, you are prompted to stop all ShoreTel voice services before running the batch update.

You can find a set of users based on any or all of the following criteria:

- Server
- User Group
- Personal Assistant
- Home Site
- Home Switch

You can globally change one of the following parameters for the selected users:

- Server
- User Group
- E-mail Domain Name
- Personal Assistant (for all call handling modes)
- Default Trunk Access Code
- Collaboration Server

To use the Batch Update Utility do the following:

Step 1 Launch ShoreTel Director with administrator privilege.

Step 2 Click **Administration > Users > Batch Update Utility**. The Batch Update Utility page appears as shown in Figure 10-22.

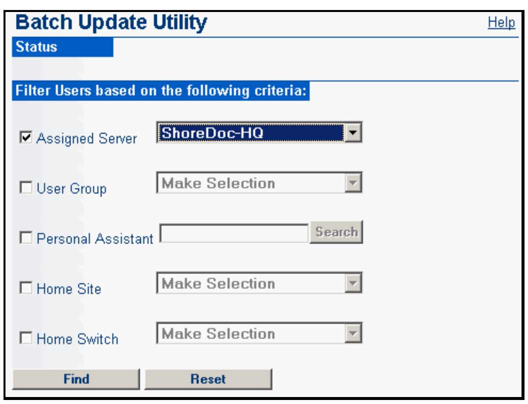


Figure 10-22 Batch Update Utility Page

Step 3 Check the check box for criteria you want to use.

Step 4 In the criteria field, select the parameters that you want to use.

Step 5 Repeat Step 3 and Step 4 for each criteria you want to use.

Step 6 Click **Find**. The Batch Update Utility Update page updates with user information as shown in Figure 10-23.

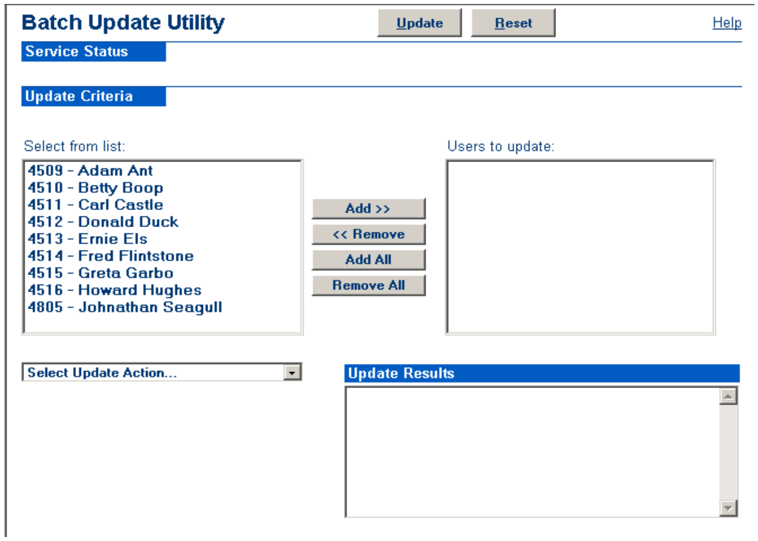


Figure 10-23 Batch Update Utility Update Criteria Page

Step 7 In the Select from list field, highlight the users you want to change and click the Add button. The users are moved to the Users to update field as shown in Figure 10-24.

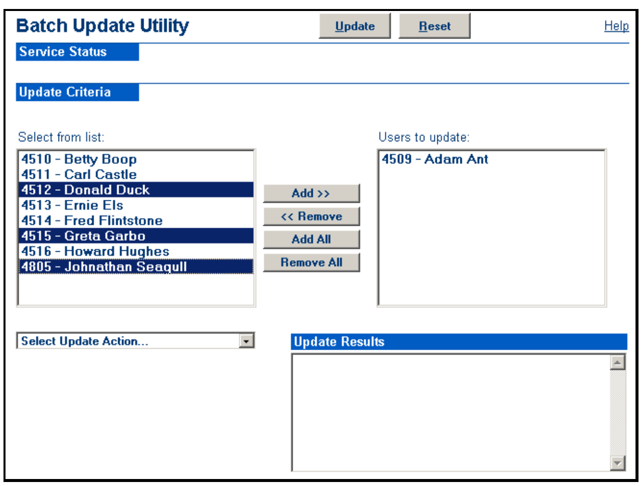


Figure 10-24 Batch Update Utility Page—Select Entities for Action

Step 8 In the Select Update Action field, select the change type that you want to implement as shown in Figure 10-25.

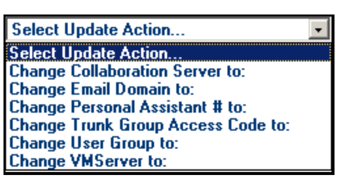


Figure 10-25 Batch Update Utility Page—Select Update Action

Step 9 Click the **Update** button at the top of the page to initiate the change. Feedback on the action is presented in the Update Results field as shown in Figure 10-26.

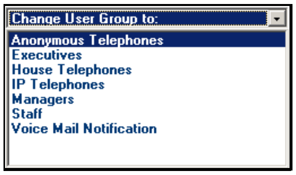


Figure 10-26 Batch Update Utility Page—Results List

When the change is complete, you can make additional batch updates.

Remember to restart the voice services when you have finished making your batch changes.

10.6.3 Notify Users

The Notify Users page, shown in Figure 10-27, enables you to notify users that their ShoreTel client – Communicator or Communicator for Mobile – has been installed or upgraded. Once the user receives notification that their client application has been installed, the user can begin configuring personal options and use the application.

To invoke automatic notification, do the following:

Step 1 Launch ShoreTel Director with administrator privilege.

Step 2 Click **Administration -> Communicator -> Notify Users**. The Notify Users page appears as shown in Figure 10-27.

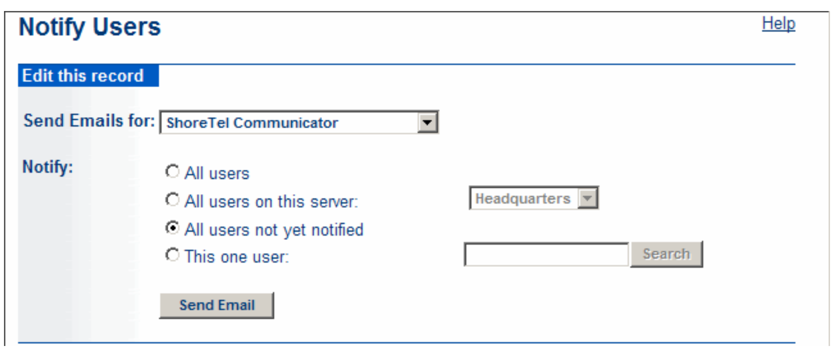


Figure 10-27 Notify Users Page

Step 3 In the **Send Email for** field, select the ShoreTel client to which you want to send notification.

Step 4 In the Notify section, select the option that you want to initiate. The options include the following:

- Click the **All users** radio button to notify all users using the selected client.
- Click the **All users on this server** radio button and in the field select the ShoreTel server to notify all users using the selected client on the ShoreTel server you select.

- Click the **All users not yet notified** radio button to notify all users using the selected client who have not been notified.
- Click the **This one user** radio button and use the Search field to select a specific user to notify to upgrade their client.

Step 5 Click **Send Email** to send the notification e-mail.

10.6.4 Extension Lists

The Extension Lists page lets you create user groups that can be used by group paging and departmental auto-attendant. To create an extension list, do the following:

Step 1 Launch ShoreTel Director with administrator privilege.

Step 2 Click **Administration -> Users -> Extension Lists**. The Extension Lists page appears as shown in Figure 10-28.

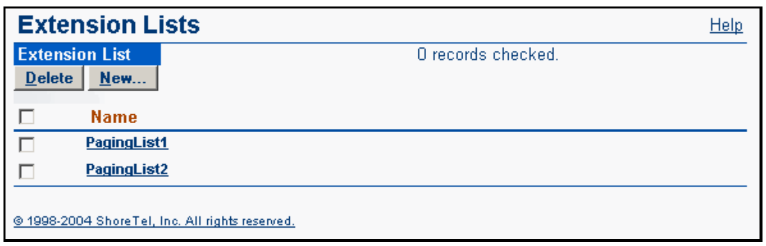


Figure 10-28 Extension Lists

Step 3 Click New to create a new list. The Edit Extension Lists page appears as shown in Figure 10-29.

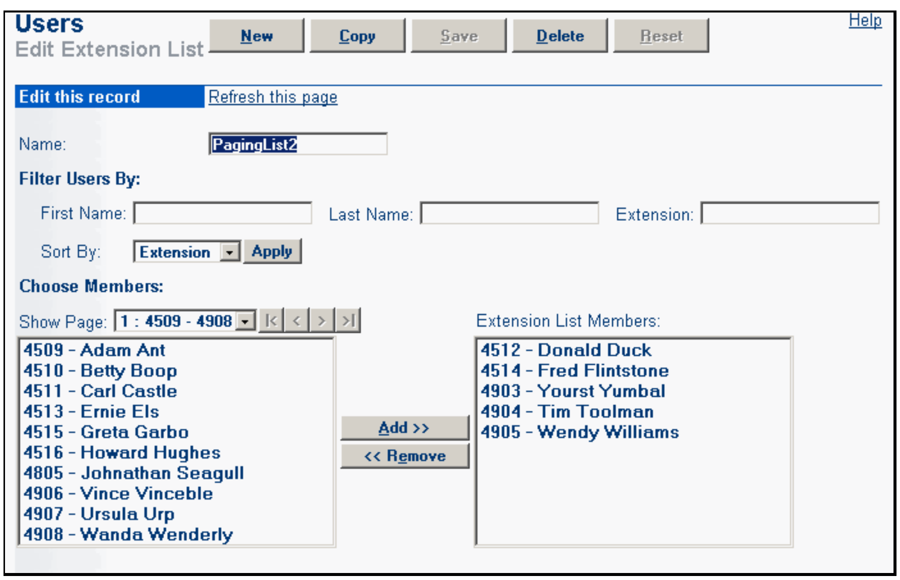


Figure 10-29 Edit an Extension List

Step 4 In the Name field, enter the name that you want to use for this extension list.

Step 5 In the Filter Users By section, do either of the following to specify how you want to sort users to include in this extension list as follows:

- In any of the filter fields (First Name, Last Name, and Extension) enter the initial characters or numbers on which you want to search.
- In the Sort field, select the criteria that you want for listing the directory.

Click **Apply**. The sorted list appears in the Choose Member field. The field lists up to 50 members.

NOTE The Show Page field groups the member entries into segments. You can use this field to select segment that you want to appear in the field.

Step 6 Select the members that you want to include in the extension list and click **Add**. The members appear in the Extension List Members field.

Step 7 Click **Save**.

CHAPTER 11

Configuring Client Applications

This chapter describes ShoreTel applications that users access during their daily activities. This chapter assumes that ShoreTel communicator is already installed on the client computer. For information about client-computer hardware, operating system, and software requirements and installation instructions, refer to the ShoreTel 12 Planning and Installation Guide. Topics include:

- "ShoreTel Communicator for Windows" on page 361.
- "Contact Center Integration with Communicator" on page 380.
- "Communicator for Mobile" on page 393.

11.1 ShoreTel Communicator for Windows

This section describes the system resource requirements and the installation steps for the ShoreTel Communicator client application.

Through its graphical user interface, an end user can manage phone calls, voice mail, and personal system settings. The ShoreTel Communicator application supports five access types and related feature sets. The five types of access are:

- Personal access
- Professional access
- Workgroup Agent access
- Workgroup Supervisor access
- Operator access

Except for Personal access, the ShoreTel Communicator features require additional licenses or specific administrative authorization. For example, ShoreTel SoftPhone and Standard Resolution Video are covered by the Professional access license. For a complete list of Communicator features, see the *ShoreTel Communicator for Windows Guide*.

The server platforms that can support ShoreTel Communicator are:

- Windows 2008 Terminal Server, 32-bit, SP2
- Windows 2008 Terminal Server, 64-bit (not R2)

11.1.1 Installation

This section describes the following topics for the installation of Communicator:

- Installation on a 64-bit Windows server
- Particular steps that are unique to deployment in a large environment that uses Microsoft Corporation's Active Directory
- Installation of Communicator for Windows

- Microsoft Office Communication Server (OCS)
- ShoreTel Converged Conference Bridge Server

11.1.1.1 64-bit Platform Supports

The 64-bit platforms on a terminal serve ShoreTel Communicator supports are Window 2003 and Windows 2008 R2. To install ShoreTel Communicator:

Step 1 Install .Net Framework for 64-bit Windows machines – Version 3.5 or higher.

NOTE .Net Framework is not supplied with the ShoreTel installation software. This software is available from Internet sources or from your software vendor.

Step 2 Install Communicator for Windows.

Step 3 Access **Control Panel > Phone and Modem > Advanced** (tab) and remove **ShoreTel Remote TAPI Service Provider**.

Step 4 Run **TSPInstall.exe**.

TSPInstall.exe is provided with the ShoreTel installation software.

Installing Communicator for Windows on a 64-bit platform places files in the following folders:

- The default location is `C:\Program Files (x86)`
- The location of 32-bit client dll files is `C:\Windows\SysWow64`

11.1.1.2 Installation in a Large Deployment that Uses Active Directory

Some unique requirements exist for the installation of ShoreTel Communicator in a large deployment that uses Active Directory. This section outlines both the manual and the automated installation of ShoreTel Communicator in a large site with Active Directory. The decision for how to proceed depends on the following key factors:

- Whether manual (local) or remote installation is used
- Presence on the server of .NET Framework version 3.5 or higher
- Access to the World Wide Web (if .NET Framework 3.5 is not on the server)

The version of .NET Framework that is used should be the 32-bit version for a 32-bit server or the 64-bit version for a 64-bit server.

Manual Installation

Manual installation of .NET Framework can be in one of two scenarios. In one scenario, the *Prerequisites* folder contains .NET Framework 3.5 (or higher). In the other scenario, .NET Framework 3.5 does not exist at all on the ShoreTel server. If the required version of .NET Framework does not exist on the system, Web connectivity is required during installation.

Manual Setup with .NET Framework in the Prerequisites Folder

The ability to install ShoreTel software manually depends on the availability of .NET Framework 3.5. If the Prerequisites folder contains .NET Framework 3.5 (along with other needed files), the system administrator can run *setup.exe* while the system is disconnected from the Web. If the system does not already have .NET Framework, the system must have an operational connection to the Web.

Manual Setup with .NET Framework Absent

If the Prerequisites folder does not contain .NET Framework 3.5, the system must have Web connectivity because, when the system administrator runs *setup.exe*, the system automatically downloads .NET Framework 3.5.

Automated Setup

If the system administrator is using automated (remote) installation to set up ShoreTel Communicator in a large network that uses Active Directory, the administrator must push the following packages in the order listed below by using Microsoft Corporation's Group Policy Object (GPO) or another deployment tool:

1. Microsoft .NET Framework 3.5 (not located in the Prerequisites folder). Either .NET Framework exists on the system or the system must be connected to the Web (so that initiation of *setup.exe* causes a download of .NET Framework.)
2. Interop Assemblies (located in the Prerequisites folder) should contain Primary Interop Assemblies for 2007 and VSTO (Visual Studio Tools for Office). This resides in the Prerequisites folder.
3. Communicator (CMWin), located in the Setup folder.

11.1.1.3 Installing Communicator for Windows

To install ShoreTel Communicator for Windows, enter the following URL in a browser:

`http://<ShoreTel_server_name>/shoreware resources/clientinstall`.

NOTE Windows hotfix KB978637 must be installed prior to the installation of ShoreTel Communicator on a 64-bit Windows 7 system.

NOTE In some PC configurations, SDA drivers used in ShoreTel 12.0 Communicator for Windows may disable Microsoft Windows 7 Aero during Web conferences active presentation share or during installation.

11.1.1.4 Installing Communicator for Mac

See the ShoreTel Communicator for Mac Client User Guide for more details about the Mac client.

ShoreTel Communicator support all Mac platforms.

Mac users download and install ShoreTel Communicator for Mac in the same way as the ShoreTel Communicator for Windows, by accessing the ShoreTel Communicator Install webpage hosted by ShoreTel Director:

http://<*serverIPaddress*>/ShoreTelresources/Clientinstall/

where <***serverIPaddress***> is the IP Address of the ShoreTel Headquarters server.

A disk image file (.dmg) is then downloaded and automatically mounted by Safari. It can then be installed on the Mac

Installation of ShoreTel Communicator for Mac is performed by dragging and dropping the ShoreTel Communicator Icon into the Mac Application folder.

11.1.2 Configuring for Instant Messaging

Instant messaging is the real-time transmission of text between two system users. Instant messages are sent from and received through a message window. Instant messages do not use call cells.

ShoreTel Instant Messaging complies with RFC 3261 – SIP: Session Initiated Protocol and RFC 3428 – Session Initiated Protocol (SIP) Extension for Instant Messaging. Instant Messaging requires ShoreTel access to Microsoft Office Communicator Server (OCS).

Instant messaging is supported by ShoreTel Communicator for Windows allowing users to interface with third party instant messaging servers that support SIP/SIMPLE protocols. ShoreTel Instant Messaging is provided to Communicator users through a ShoreTel Service Appliance 100, a ShoreTel Converged Conference Bridge Sever, or Microsoft Office Communications Server (OCS).

ShoreTel Service Appliance 100 (SA-100)

The ShoreTel Service Appliance 100 also supports ShoreTel instant messaging and presence information exchanges. Refer to the *ShoreTel Service Appliance 100 (SA-100)Planning, Installation and Administration Guide* for more information.

ShoreTel Converged Conference Bridge Server

The ShoreTel Converged Conference Bridge Server Version 7.1 supports ShoreTel instant messaging and presence information exchanges. Refer to the *ShoreTel Converged Conferencing Director Installation and Administration Guide* for more information.

Microsoft Office Communication Server

This section describes Microsoft OCS implementation requirements for compatibility with the ShoreTel system. Readers should keep in mind that Microsoft Corporation maintains some of these requirements, and compatibility can change without notice.

- **OCS Versions:** ShoreTel is compatible with the Microsoft OCS 2007 Standard edition or better.

  OCS software is available in three editions: Standard, Enterprise, and Enterprise Expanded. Standard Edition is typically sufficient for ShoreTel deployments with less than 5,000 users.

- The requirements for OCS system components include:
  - — Windows Server 2003 or later. The 2003 version is compatible with Microsoft OCS 2007.
  - — Windows Active Directory.
- License Requirements: OCS requires one standard client access license (CAL) for each ShoreTel user enabled for instant messaging.
- Server Configuration: The OCS server must be configured for UDP transport. Communicator for Windows support Transport Layer Security (TLS) protocol.
- Connectivity Testing: To test ShoreTel server connectivity, log two OCS servers to the ShoreTel server and send an instant message between the OCS servers.

- Additional Information: Additional OCS information is available from the following sources:
  - — The Office Communicator 2007 Resource Center at:

    http://office.microsoft.com/en-us/help/HA102373941033.aspx
  - — “Microsoft OCS Requirements for ShoreTel Communicator Instant Messaging Feature ST-10063.”

    These application notes are posted on the support site and searchable through the Knowledge Base.

11.1.2.1 Setting up the System Server

The following presence servers support ShoreTel Instant Messaging:

- horeTel Service Appliance 100 (SA-100)

  NOTE Support IM transmissions through HTTPS
- Converged Conference Bridge Server Version 7.1.

  NOTE Support IM transmissions through HTTPS
- Microsoft Office Communications Server, Version 2007.

  NOTE When using OCS, ShoreTel supports IM transmissions through TCP, TLS, and UDP.

NOTE When using the ShoreTel Converged Conference bridge, ShoreTel IM users will see the following behavior when Communicator for Windows presence invitation is set to 'Prompt to accept invitation':

- If a user (user A) sends an IM to another user (user B) and neither user has granted permission to share presence information with the other, the Communicator will only prompt the IM recipient (user B) to share presence information, but not the sender (user A).
- The sender's IM presence (user A) will be displayed in an 'Away' (Yellow) status until user B sends an IM to user A, prompting a request to share presence information.

ShoreTel permits multiple presence servers on a system. Each user can be assigned to a maximum of one presence server. Users can exchange instant messages only with other users assigned to the same presence server.

To associate a presence server with the ShoreTel system:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Application Servers > IM Servers**. The IM Server List page appears as shown in Figure 11-1.

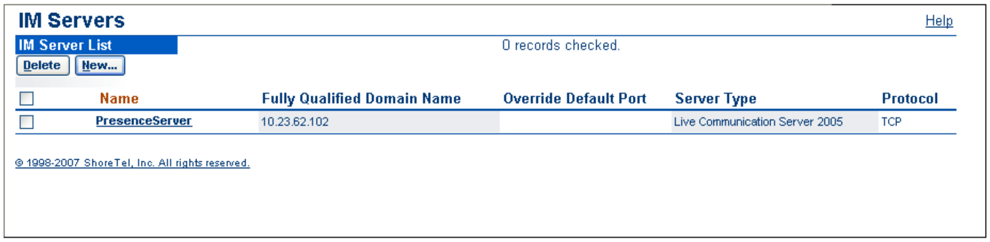


Figure 11-1IM Server List Page

Step 3 Click the **New** button. The IM Server Info dialog box appears as shown in Figure 11-2.

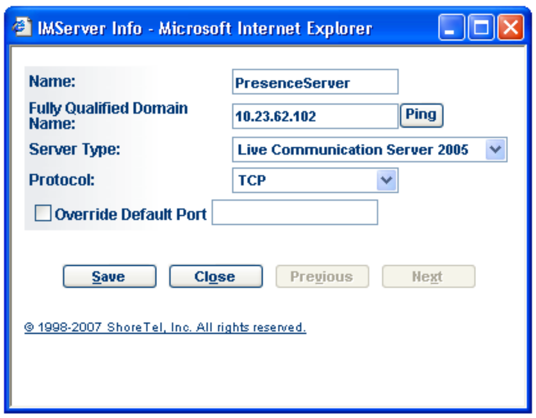


Figure 11-2IM Server Popup

Step 4 In the Name field, enter the name that you want to use for this IM profile.

Step 5 In the Server Type field, select the type of server that you want to use for IM.

Step 6 In the Protocol field, enter the protocol the IM server uses.

Step 7 In the Host field, enter the domain or IP address of the IM device you want to use to provide IM services for this profile.

Step 8 Check the **Override Default Port** checkbox if the server is using a port other than 5222.

Step 9 Press the **Save** button in the IM Server popup.

Step 10Ping the IP address entered in the IMServer Info page to verify that the address is accessible.

When changing the Presence Server version, access the IM Server popup and enter the presence version in the Server Type data entry field.

11.1.2.2 Configuring a User for Instant Messaging

The following Communicator types can send and receive Instant Messages:

- Professional Access
- Workgroup Agent Access
- Workgroup Supervisor Access
- Operator Access

Configuring Instant Messaging for a User is a two step process:

Step 1 The administrator enables instant messaging for the user.

"Enabling Instant Messaging for a System User" on page 367 describes the process of enabling instant messaging for a user.

Step 2 The user configures instant messaging user options.

The *Communicator for Windows User Guide* has instructions for configuring instant messaging options in the user's Communicator window.

11.1.2.3 Enabling Instant Messaging for a System User

To enable Instant Messaging for a system user:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Users > Individual Users**. The Individual User List page appears.

Step 3 In the First Name column, select the user that you want to enable. The Edit User page appears as shown in Figure 11-3.

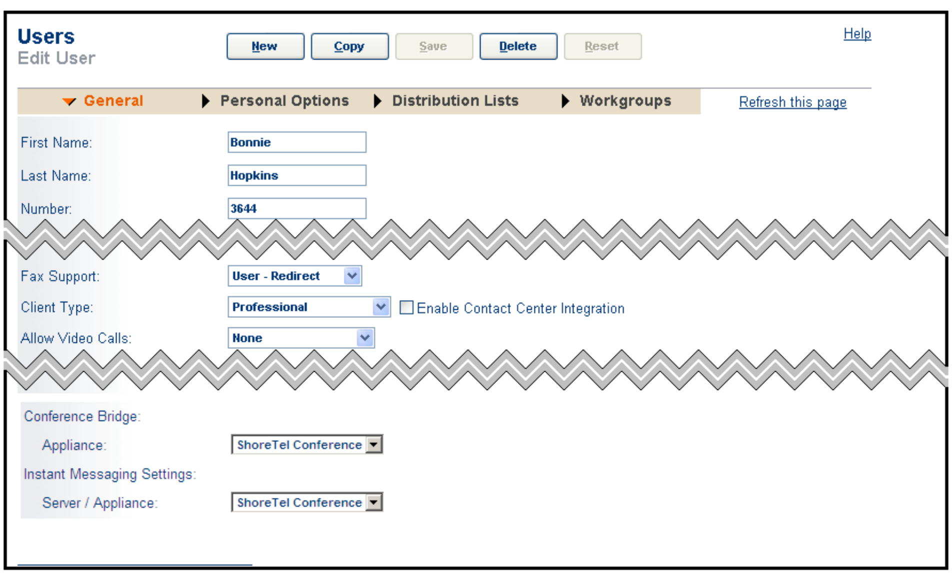


Figure 11-3Instant Messaging Parameters on the Edit User Page

Step 4 Verify the First Name and Last Name parameters list the user for which you are enabling Instant Messaging.

Step 5 In that Access License field, select a license for the use that supports IM: Professional, Workgroup Agent, Workgroup Supervisor, or Operator.

Step 6 Check the **Allow Telephony Presence** check box.

Step 7 Scroll down to the Instant Message Settings section and in the Server/ Appliance field, select the SA-100 or other IM server that you want this user to use.

Step 8 In the IM ID data entry field, enter the address through which the user will receive and send Instant Messages as it is configured on the OCS server.

NOTE This field does not appear and is not required if you are using an SA-100 for instant messaging.

Step 9 Press the "Save" button at the top of the page to enable the changes.

Step 10 Verify that the presence server is configured to authorize access by the address specified for the user in Step 8.

Consult the user documentation for the presence server for instructions on authorizing user access.

Step 11 Do the following to advise the user to verify access to the presence server:

Step a Opening the "Options and Preferences" page from the user's Communicator Main Window.

Step b Selecting Instant Messaging in the Menu on the left side of the page.

Step c Entering the user's OCS User Name and Password in the Account Information section at the top of the page.

Step d Pressing the Reconnect button.

The Status Bar in the Main window will display the results of Communicator's attempt to access the presence server.

11.1.3 Presence

Presence is a ShoreTel feature that identifies, uses, and distributes the availability of system users and other personal contacts. Presence information allows users to verify the availability of other users before attempting to contact them. Presence improves overall enterprise productivity by reducing calls to unavailable parties and by providing the enhanced ability to immediately schedule meetings, events, and communication sessions based on the availability of desired participants. Users can also choose when to receive Instant Messages by changing their presence settings.

Communicator for Windows defines three presence settings for each user.

- **Telephony presence** indicates a user's availability to accept voice calls.
- **Instant Messaging presence** indicates a user's availability to engage in IM conversations.
- **Combined presence** is a single setting that indicates user availability on the basis of Telephony and IM presence status.

Communicator automatically adjusts the presence status of users as they use system resources. Users can also manually adjust their status at any time.

A user can monitor the presence of a maximum of 500 other contacts. Administrators configure the number of contacts a user can monitor by setting the maximum contact list size, as described in the "Setting the Maximum Contact List Size" on page 369.

11.1.3.1 Configuring a User for Presence

Telephony presence is available to users of all Communicator types. IM presence is available to users authorized to use Professional, Operator, Workgroup Agent, or Workgroup Supervisor Communicator. Director settings control user access to both presence types.

Configuring Presence for a User is a two step process:

Step 1 The administrator enables presence for the user.

The "Enabling Presence for a User" on page 369 describes the process of enabling presence for a user.

Step 2 The user configures presence user options.

Refer to the Communicator for Windows User Manual for instructions on configuring presence user options from the user's Communicator window.

11.1.3.2 Enabling Presence for a User

To enable a user for telephony presence:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Users > Individual Users.** The Individual Users page appears.

Step 3 In the First Name column, click on of the desired user. The User page appears.

Step 4 Verify the First Name and Last Name parameters list the user for which instant messaging is being enabled.

Step 5 Check the **Allow Telephony Presence** check box.

Step 6 Click **Save**.

To enable a user for IM presence:

Users that are authorized for instant messages are automatically configured for IM presence.

11.1.4 Setting the Maximum Contact List Size

Contacts are directory entries whose information is organized into specialized lists called Contact Lists to provide convenient access by the Communicator user. Contacts are selected from directories accessible to Communicator, including the System Directory, Personal directory or Microsoft Outlook. Contact Lists, which are normally accessed from the Communicator main page, provide quick access to regular directory entries without navigating through the directory menus. Users can perform tasks on Contact List entries that are available from other directories such as initiating voice call or instant messages, handing active calls, and sending email or voice messages.

The maximum ShoreTel Contact List size is 500 contacts for each user. The size of each user's Contact List is controlled through a Class of Service setting.

To set the maximum Contact List size for a selected Class of Service:

Step 1 Open the Class of Service list page by selecting ***Administration -> Users -> Class of Service*** in the Director Menu.

Step 2 Open the Edit Telephony Features Permissions page by clicking the name of the selected Telephony Features Permissions class of service at the top of the Class of Service list page.

Step 3 Enter the maximum Contact List size in the Max. Buddies Per User field located at the top of the Edit Telephony Features Permissions page, as shown in Figure 11-4.

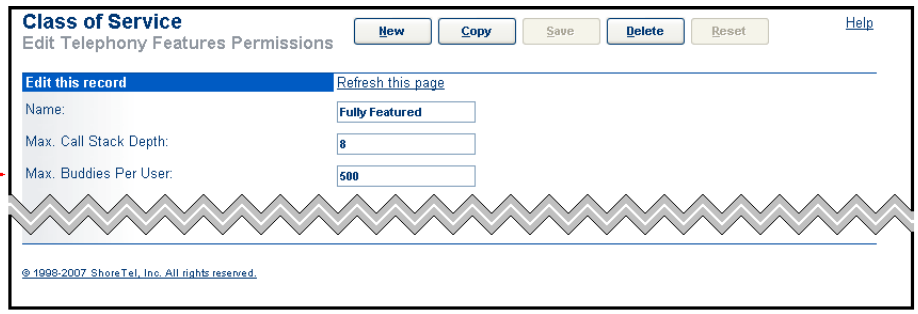


Figure 11-4Setting the Maximum Contact List Size for a Class of Service

The maximum Contact List size specified by a class of service is imposed upon a user when the class of service is assigned to that user.

11.1.5 Enable SoftPhone for Users

SoftPhone is a completely integrated software component of Communicator for Windows. SoftPhone users participate in voice calls and listen to messages through a headset or speakers with a microphone without any additional hardware.

The **Edit User – General** page in Director enables users to access to SoftPhone.

To enable SoftPhone access for a selected user:

Step 1 Open the Individual User list page in Director by selecting **Administration -> Users -> Individual Users** from the main menu.

Step 2 Open the Edit User page by clicking on the First Name of the desired user in the Individual User List.

Step 3 Verify the First Name and Last Name parameters list the user for which SoftPhone service is to be enabled.

Step 4 Verify that the General menu tab is selected on the page selection bar at the top of the page.

Step 5 Select the **Allow Use of SoftPhone** parameter, located in the center of the page.

Step 6 Press the SAVE button at the top of the page to enable the changes.

11.1.6 Video Calls

ShoreTel supports video calls between two Communicator users that are communicating directly. However, Communicator instances cannot have video when these instances are communicating through a ShoreTel server. ShoreTel does not support server-based, star-configurations or interoperability with other video solutions. ShoreTel supports video calls through an SVC codec.

ShoreTel does not support video-only calls. Video is added to a voice call when requested by a user. Video can either be added to an active voice call or established as a part of a new voice call.

IMPORTANT *End users who attempt video calls must have the most recent video driver and have all the updates installed for the video card. Refer to the documentation from the manufacturer of the individual video card for complete information.*

11.1.6.1 Supported Video Cameras

For a list of Video Cameras currently supported by Communicator, refer to “Video Camera Requirements for ShoreTel Communicator, Feature AN-10062”. These application notes are posted on the support site and searchable through the Knowledge base.

11.1.6.2 Enabling Video Calls for Users

The Edit User – General page in Director enables users for video calls.

To enable Video Call access for a selected user:

Step 1 Open the Individual User list page in Director by selecting **Administration > Users > Individual Users** from the main menu.

Step 2 Open the Edit User page by clicking the First Name of the desired user in the Individual User List.

Step 3 Verify the First Name and Last Name parameters list the user for which Video is to be enabled.

Step 4 Verify that the General menu tab is selected on the page selection bar at the top of the page.

Step 5 Select the desired option from the Allow Video Calls drop-down menu located in the center of the page.

The drop-down menu options include

- **None:** The user cannot make video calls or add video to existing voice calls.
- **Standard Resolution:** The user can perform VGA resolution video calls.

Step 6 Press the Save button at the top of the page to enable the changes.

11.1.6.3 Enabling Intersite Video Calls

The Video call site setting is a Class of Service parameter. Members of a class of service can only conduct video calls with other Communicator users at the same site unless this parameter is set for intersite calls.

To enable intersite video calls for a class of service:

Step 1 Open the Class of Service list page by selecting **Administration > Users > Class of Service** in the Director Menu.

Step 2 Open the Edit Telephony Features Permissions page by clicking the name of the selected Telephony Features Permissions class of service at the top of the Class of Service list page.

Step 3 Select the **Allow Intersite Video Calls** parameter on the top portion of the Edit Telephony Features Permissions page.

11.1.7 Programming Personal Communicator Toolbar

Programmable Toolbars for Communicator are similar to regular toolbars in that they extend across the top of the Communicator user interface. However, the buttons in these new toolbars can be programmed with common operations in a manner similar to the programmable buttons on some IP phone models.

Once a user's toolbar buttons have been configured, the user can perform many basic telephony operations just by clicking a button in Communicator. For example, a button could be configured to "speed dial" another extension or open up the user's default browser to a programmed URL when clicked.

In addition, Programmable Toolbars also provide the foundation for integrating ShoreTel Communicator with Contact Center's Agent Toolbar. End users can assign Contact Center functions to the buttons on the Communicator Toolbar and control their contact center state while accessing call control functions from a single, unified interface.

Details:

- Toolbars must be configured by the system administrator via ShoreTel Director.
- Toolbars can be configured per-user (see Table 11.1.7.1 on page 375) or globally (see the "Creating a Global Programmable Toolbar" on page 377).
- Up to 6 toolbars can be defined per user, with each toolbar supporting up to 24 programmable buttons.
- Each user can additionally inherit up to 3 global toolbars through their user group.
- Each toolbar can exist on a separate row in the UI or toolbars can share a row.
- Rows can be docked or moved around the top portion of the window as a group.
- Individual toolbars can be shown or hidden from the View menu in the Communicator user inter-face (see Displaying Toolbars on page 72).
- Buttons may combine an operation with a parameter (such as a user extension), allowing one-click access to commonly performed operations, such as blind-transferring to a particular user.
- Buttons assigned with a particular operation will be disabled when their corresponding menu items are disabled. For example, some of the Contact Center function buttons are not available as menu items, thus ensuring that the system administrator can easily block access to those functions on a per-user basis.
- Buttons that are associated with an extension (such as Blind Transfer or Extension Monitor) will show the Communicator-user when the monitored party has call activity. Additional presence information appears if the user hovers the cursor over the associated button.
- The user can have any and all toolbars active at once.

- Supported operations are listed on the following page.

Supported operations:

- Add/Modify Contact
- Agent Login
- Agent Logout
- Agent Wrap-Up
- Answer
- Answer Call Center Call
- Assign to Last External Number (Extension Assignment / Extension)
- Barge In
- Blind Transfer Agent
- Bridged Call Appearance
- Change CHM
- Change Default Audio Path
- Conference
- Conference Blind
- Conference Consultative
- Conference Intercom
- Consult Transfer Agent
- Dial Mailbox
- Dial Number (Speed Dial)
- Edit Call Note
- End Wrap-Up
- Execute DDE command
- Extend Wrap-Up
- Go Home
- Group Pickup
- Hangup
- Help
- Hold
- Intercom
- Invoke Command line
- Invoke URL
- Login Group
- Login Primary Groups

- Logout Group
- Logout Primary Groups
- Monitor Extension
- Open Agent Monitor
- Open Conference Mgr
- Open Control page
- Open Directory
- Open Extension Monitor
- Open External Assignment
- Open History Viewer
- Open Queue Monitor
- Open Soft Phone
- Open Voice Mail
- Page
- Park
- Park and Page
- Pickup
- Pickup Night Bell
- Record Call
- Record Extension
- Reinsert Busy Call
- Reinsert Terminated Call
- Reinsert Unanswered Call
- Release with Code
- Resume/Release
- Run Contact Center App
- Send Digits Over Call
- Set Agent ID
- Silent Monitor
- Supervisor Help
- To AA
- To VM
- Toggle Handsfree
- Transfer
- Transfer Blind

- Transfer Consultative
- Transfer Intercom
- Transfer to Mailbox
- Transfer Whisper
- Unpark
- Whisper Page
- Whisper Page Mute
- Wrap Up Code

11.1.7.1 Creating a Personal Programmable Toolbar

To configure the buttons of a user's Communicator Programmable Toolbar, follow the instructions below:

Step 1 Launch ShoreTel Director and enter the user ID and password.

Step 2 Click **Administration > Users > Individual Users**. The Individual User page appears.

Step 3 Select the user whose profile you want to edit. The Edit User page appears.

Step 4 Click the **Personal Options** tab. The Personal Options tab appears as shown in Figure 11-5.

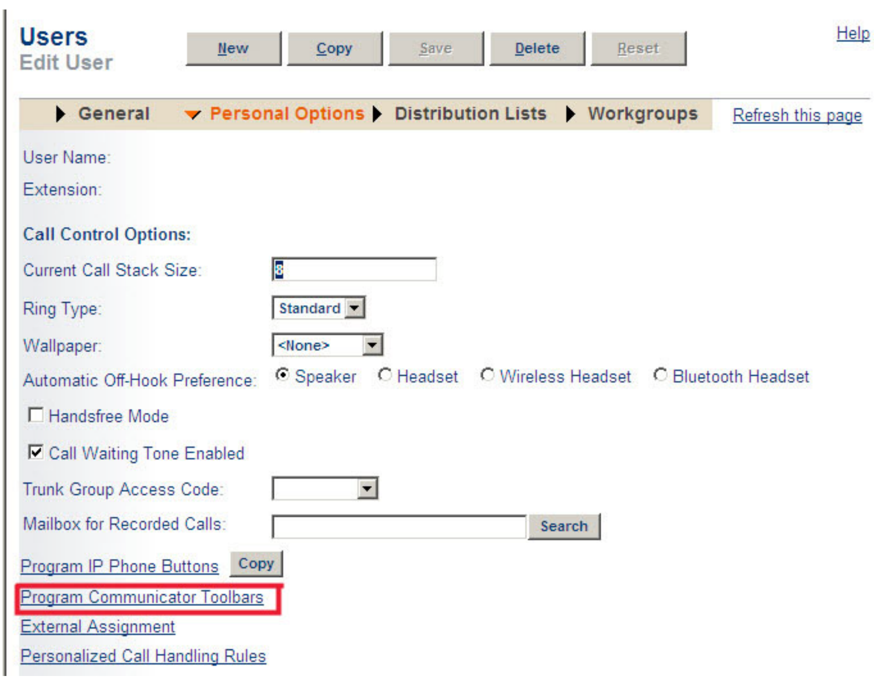


Figure 11-5Personal Options Tab

Step 5 Click the **Program Communicator Toolbars** link. The Program Communicator Toolbars page appears as shown in Figure 11-6.

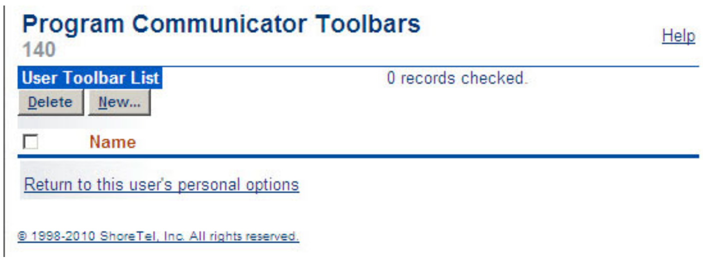


Figure 11-6Program Communicator Toolbars Page

Step 6 Click the **New** button. The Program Communicator Toolbar Buttons page appears as shown in Figure 11-7.

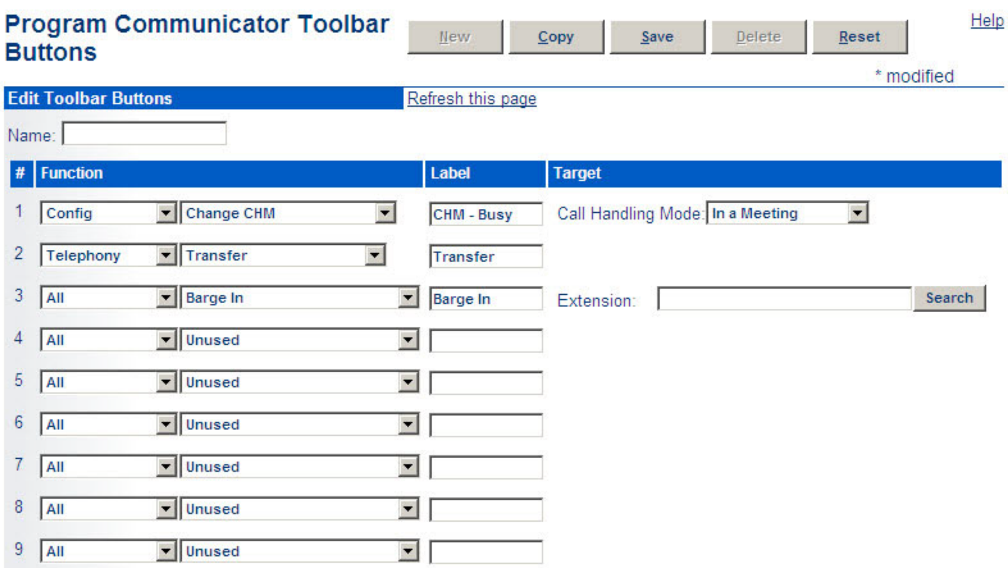


Figure 11-7Program Communicator Toolbar Buttons

Step 7 In the **Name** field, enter the name you want to use for this toolbar.

Step 8 Click on the first Function drop-down menu (which should say "All") and select the category. Options are:

- **All -** This lists all operations
- **Contact Center -** (For example, Login Group, Logout Group, Answer Call Center Call, etc.)
- **Config -** (For example, toggle Handsfree mode, toggle audio path, Agent Wrap-Up, etc.)
- **Other -** (For example, Unused)
- **Telephony -** (For example, Conference, Intercom, Group Pickup, etc.)
- **Windowing -** (For example, Open History Viewer, Open Agent Monitor, Open Control Panel, etc.)

Step 9 Click on the second ***Function*** drop-down menu (which should say "Unused") and select the specific operation this button identifies.

Step 10 Enter a label in the ***Label*** field. The maximum is 12 characters.

Step 11 In the Target field, enter the appropriate information as required for the type of operation the button will be performing. (For example, if the desired operation is changing the CHM, then a drop-down menu would appear in the Target area, allowing you to select from the various modes.)

Parameters are mandatory for some functions (e.g. Change CHM) and are optional for others (e.g. Blind Transfer). For optional parameters, the user is prompted to provide information for the optional parameter the first time the function executed via the programmable toolbar.

Step 12 Continue programming the toolbar buttons as desired.

Step 13 If a button is left unused or blank and this unused button is between other used buttons, Communicator will interpret the sequence of blank buttons as a divider that will be visible on the toolbar.

Step 14 Once you have finished configuring the toolbar programmable buttons, click **Save** to store your changes.

When finished programming the toolbar for a user, the user's Communicator window should now have a toolbar similar to the one shown in Figure 11-8.

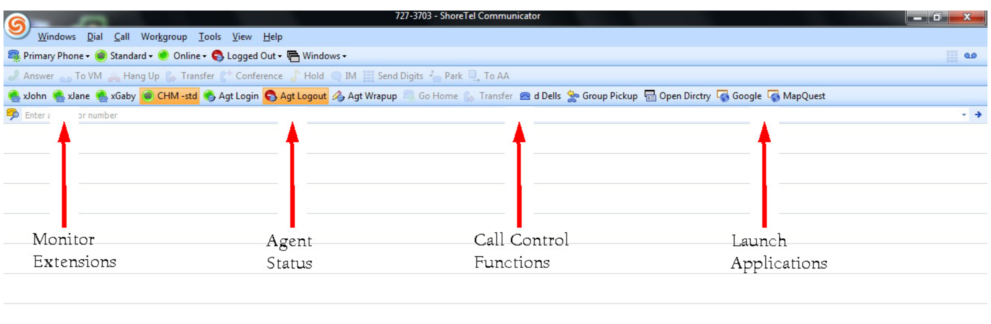


Figure 11-8 Communicator Window with Programmable Toolbar

11.1.7.2 Creating a Global Programmable Toolbar

Alternatively, instead of configuring the Programmable Toolbar for an individual user, you can configure a global Programmable Toolbar which can then be applied to many users at once, thus simplifying the configuration process.

To create a global Programmable Toolbar, follow the instructions below:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Communicator > Global Toolbars**. The Global Toolbars List page appears as shown in Figure 11-9.

Figure 11-9Global Toolbar List Page

Step 3 Click the **New** button. The Edit Global Toolbar Buttons page appears as shown in Figure 11-10.

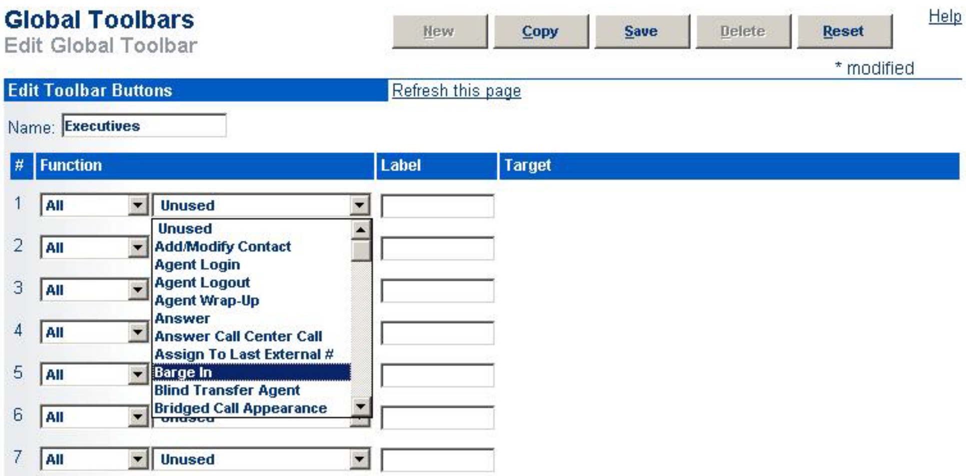


Figure 11-10Edit Global Toolbar Page

Step 4 In the Name field enter a name for this toolbar.

Step 5 Click the first Function field (All is the default) and select a category:

- **All** - This lists all operations.
- **3rd Party** - (For example, Login Group, Logout Group, Answer Call Center Call, and so on).
- **Config** - (For example, toggle Handsfree mode, toggle audio path, Agent Wrap-Up, and so on).
- **Other** - (For example, Unused).
- **Telephony** - (For example, Conference, Intercom, Group Pickup, and so on).
- **Windowing** - (For example, Open History Viewer, Open Agent Monitor, Open Control Panel, and so on).

Step 6 Click the second Function field (which should say "Unused") and select the specific operation this button will perform.

Step 7 Enter a label in the Label field. The maximum size is 12 characters.

Step 8 In the Target field, enter the appropriate information as required for the type of operation the button will be performing. (For example, if the desired operation is changing the Call Handling Mode, then a Call Handling Mode drop-down menu appears in the Target area, letting you select from the various CHMs.)

Parameters are mandatory for some functions (e.g. Change CHM), while they are optional for others (e.g. Blind Transfer). For optional parameters, the user is prompted to provide information for the optional parameter the first time the function executed via the programmable toolbar.

Step 9 Continue programming the toolbar buttons as needed.

If a button is unused or blank and this unused button is between other, used buttons, Communicator shows the sequence of blank buttons as a divider that is visible in the toolbar.

Step 10 Click **Save** to store the changes.

To assign a global programmable toolbar to a user profile, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > User > User Groups**. The User Groups page appears.

Step 3 Select select the user group to which you want to assign a global programmable toolbar. The Edit User Group page appears as shown in Figure 11-11.

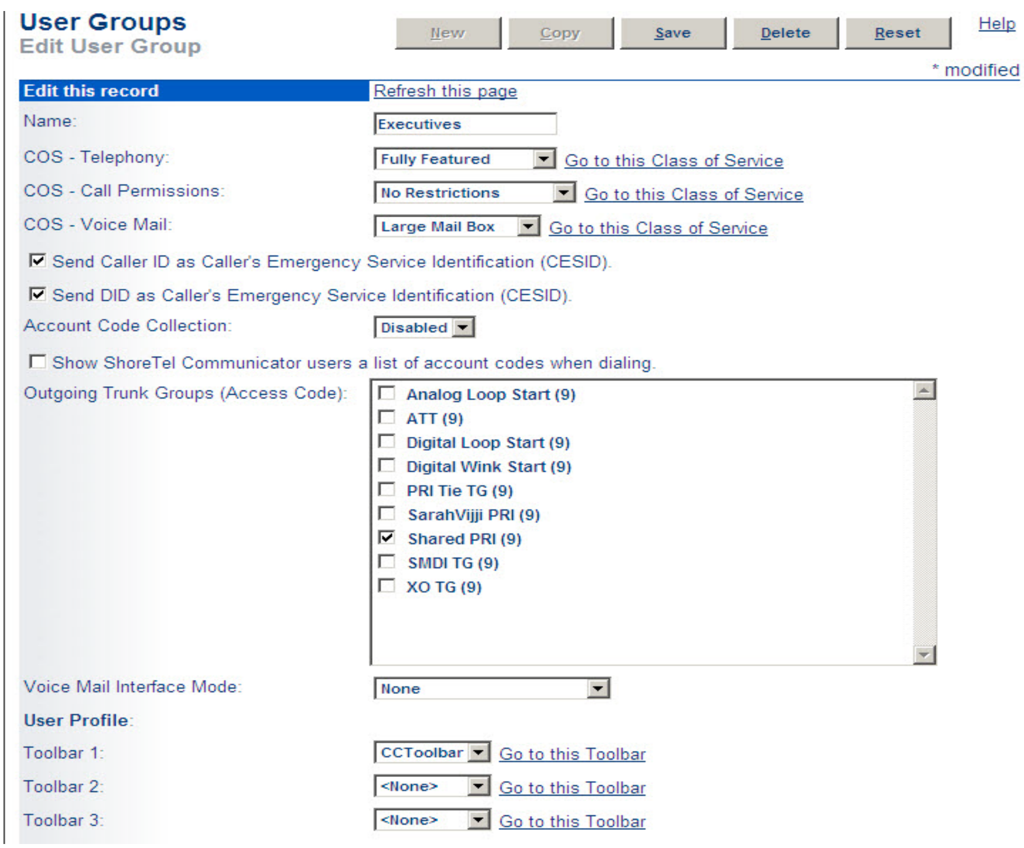


Figure 11-11 Edit User Group Page

Step 4 In the User Profile section, click the Toolbar 1 field and select the desired global toolbar. All members of this user group will now have this toolbar appear on their Communicator.

Step 5 Repeat Step 4 if multiple toolbars are desired - up to three global toolbars can be assigned to a user group.

Step 6 Click **Save** to store the changes.

11.2 Contact Center Integration with Communicator

11.2.1 Description

ShoreTel integrates Contact Center functionality into Communicator, allowing end users to control their contact center state and access Contact Center functions and Communicator operations from a single, unified interface.

Figure 11-12 displays the non-integrated Contact Center Agent Toolbar.

Figure 11-12Contact Center Agent Toolbar

This interface allows Contact Center users to program buttons on the toolbar by assigning common functions and call center operations to the various buttons. Users can perform the desired operation with the click of a button.

Contact Center functions are integrated into Communicator by assigning Contact Center operation buttons into Programmable Toolbars, as described in the "Programming Personal Communicator Toolbar" on page 372. Each toolbar can contain 24 buttons. To display additional buttons beyond those normally visible, click the arrow icon on right side of the toolbar. Figure 11-13 displays a Communicator toolbar that contain Contact Center function buttons.

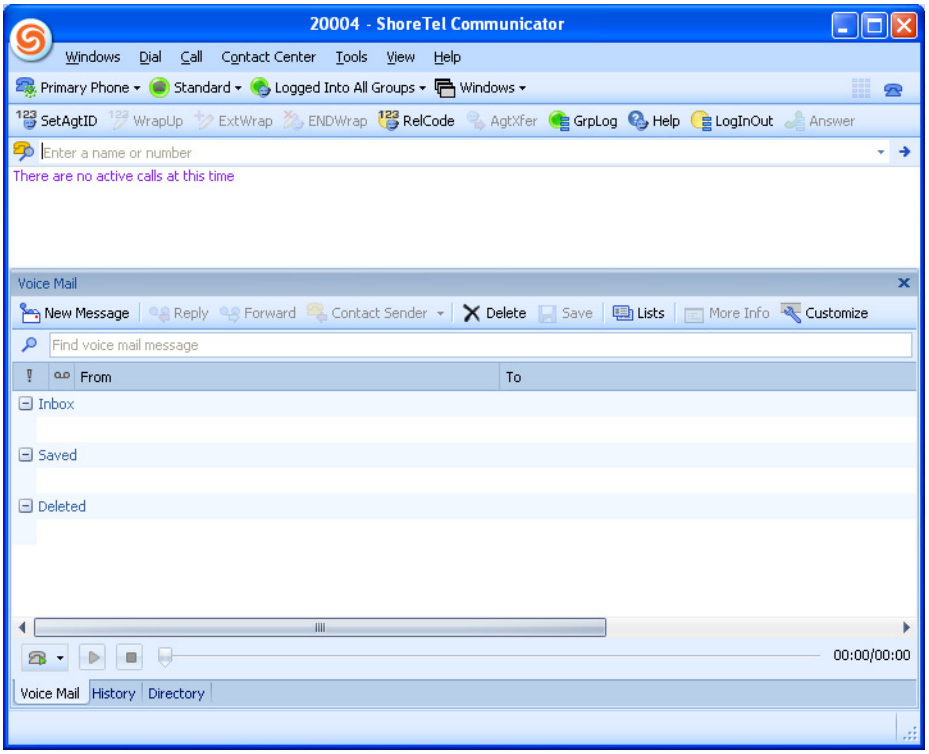


Figure 11-13Communicator Toolbar Containing Contact Center Operation Buttons

Details:

- Up to 24 buttons can be programmed in a single Communicator toolbar.
- Toolbars can be created on a per-user basis or on a global basis. Toolbars created at the global level can be deployed to multiple users in the system.
- Three global toolbars can be used per User Group, with a maximum of 100 per system.
- Six personalized toolbars can be created for each user.
- A user can employ 3 global toolbars in addition to 6 personalized toolbars, (total of 9 toolbars). With 24 buttons per toolbar, a user can deploy a total of 216 programmable toolbar buttons. (24 buttons per toolbar X 9 toolbars = 216 buttons.)

11.2.2 Accessing Contact Center Agent Toolbar Functions

The available Contact Center operations in Table 11-1 show the name of corresponding operations in the integrated Communicator (if applicable). An operation that does not appear in the Contact Center Agent Toolbar column does not have an analogous function in the integrated Communicator. Often, a Contact Center operation does not have an analogous Communicator operation because the Contact Center operation can be done through an existing function in the regular (non-integrated) ShoreTel Communicator.

11.2.3 Configuration

To configure integrated Communicator – Contact Center for a user:

Step 1 Launch ShoreTel Director.

Table 11-1 Communicator access methods to Contact Center operations

Contact Center Operation Name	Operation Button or Path in Communicator
Telephony Operations	
Answer	Answer Call Center Call
Set Callback - Reinsert Busy	Reinsert Busy Call
Set Callback - Reinsert No Answer	Reinsert Unanswered Call
Set Callback - Reinsert Terminate	Reinsert Terminated Call
ACD Operations	
Groups Manager	Select **Contact Center > Agent Manager** from Communicator
Login Primary Groups	Login Primary Groups
Logout from Primary Groups	Logout Primary Groups
Login Group	Login Group
Logout Group	Logout Group
End Wrap Up (Ready)	End Wrap-Up
Release with Code	Release With Code
Resume/ Release	Resume/Release
Supervisor Help	Supervisor Help
Transfer to Agent	Blind Transfer Agent
Wrap-Up Code	Wrap-Up Code
Wrap-Up Manual Control	Extend Wrap-Up
Accessing Applications Windows	
Desktop Wallboard	Select **Contact Center -> Desktop Wallboard** from Communicator
Call Status	Select **Contact Center -> Call Status** from Communicator
Queue Calls	Select **Contact Center -> Queue Calls** from Communicator
Agent Log	Select **Contact Center -> Agent Log** from Communicator
Contact Center Reports	Select **Contact Center -> Contact Center Reports** from Communicator
Other Operations	
Pop-up window as client is launched	Set Agent ID
Execute an Application	Run Contact Center App. (or execute from Command Line)
Contact Center Functions Not Available in Communicator	

Table 11-1 Communicator access methods to Contact Center operations (Continued)

Contact Center Operation Name	Operation Button or Path in Communicator
Divert Incoming Call	
Manage List of Login / Logout ACD Groups	
Open Telephony Window	
Chat Tree	

Step 2 Click **Administration > Users > Individual** Users.

Step 3 Click on the name of the user whose toolbar you would like to configure. The Edit User page appears as shown in Figure 11-14.

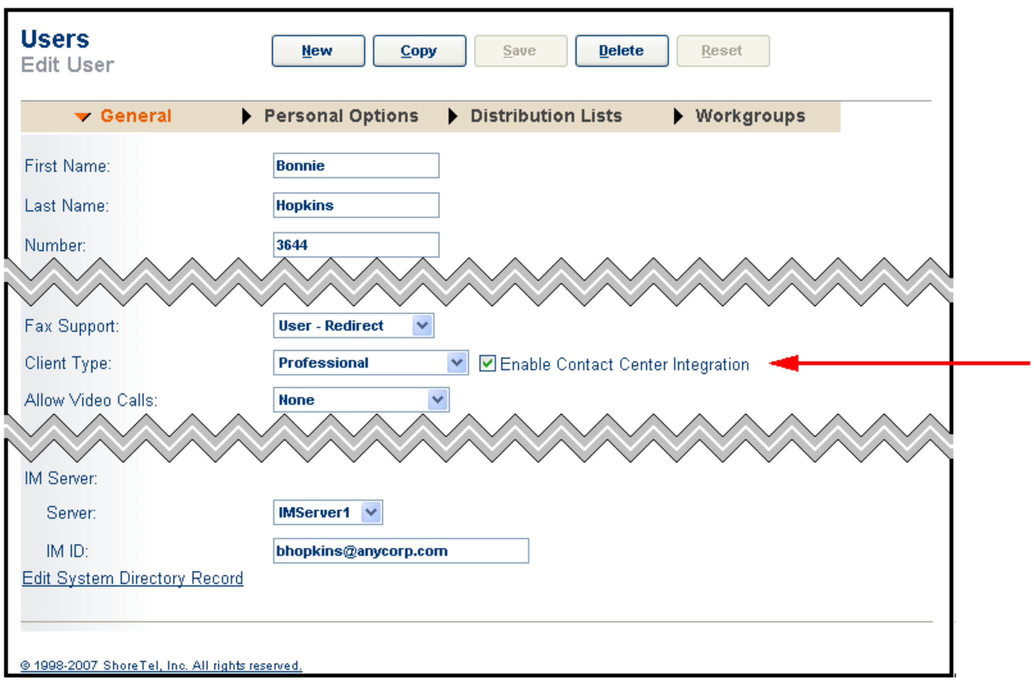


Figure 11-14Edit User Page

Step 4 In the Access License field, select a license that supports Contact Center (Personal or Professional).

Step 5 Check the **Enable Contact Center Integration** check box.

Step 6 Click Save to store changes.

To configure the programmable buttons toolbar for this user, do the following:

Step 1 Click the **Personal Options** tab for the user who's system is to be configured, and then click the Program Communicator Toolbars link.

Step 2 On the Program Communicator Toolbar page, click the **New** button. The Program Communicator Toolbar Button page appears as shown in Figure 11-15.

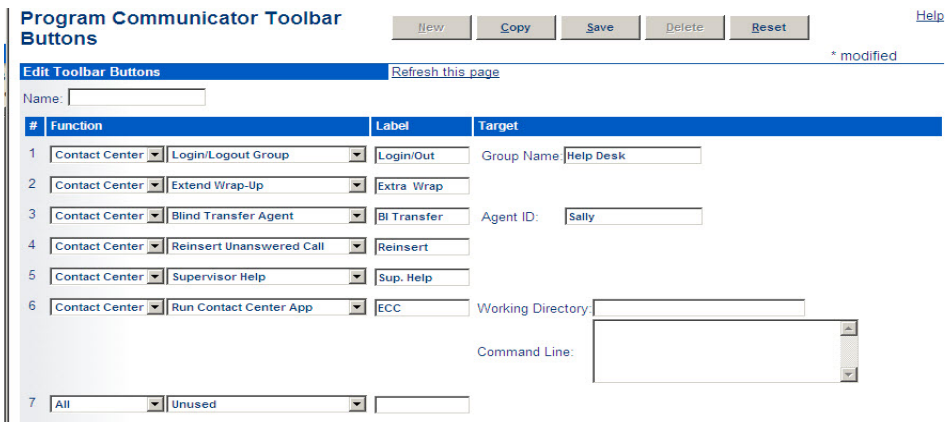


Figure 11-15Program Communicator Toolbar Button Page

Step 3 Enter a name for the toolbar and then click on the Function drop-down menu and select Contact Center.

Step 4 Then, select the desired operation from the drop-down menu. This operation will be associated with this specific toolbar button.

Step 5 Enter a label in Label field.

Step 6 If the function requires Target information, the required fields will appear to the right of the function. Enter the appropriate information as required for the type of operation the button will be performing. (For example, if the desired operation is Blind Transfer Agent, then an Agent ID field would appear in the Target area).

If a required target field for an operation is left blank or is configured with invalid information, the system will open a pop-up dialog box to collect more information (or valid information) when the user clicks on the associated button in Communicator. This behavior applies to the following operations:

- Consult Transfer to Agent
- Blind Transfer to Agent
- Login/Login Group
- Logout Group
- Release with Reason Code
- Wrap-up with Code

Step 7 Continue programming the toolbar buttons as desired.

If a button is left unused or blank and this unused button is between other used buttons, Communicator will interpret the sequence of blank buttons as a divider that will be visible on the toolbar.

Step 8 Once the configuration of the toolbar programmable buttons is complete, click Save to store the changes.

Details

- Repeat this process of assigning operations to buttons for each new user that will be using the integrated Communicator.
- When finished configuring buttons for each new user, follow the procedure below to configure the Communicator clients.

11.2.4 Configuring Call Center Options through Communicator

To configure the Contact Center options for a Communicator user, refer to the Communicator User's Manual.

11.2.5 Integrating ShoreTel Communicator with Contact Center

To configure integrated ShoreTel Communicator – Contact Center for a user:

Step 1 Launch ShoreTel Director; type the user ID and password; and then click Login.

Step 2 Click on the Administration link if necessary to expand the list.

Step 3 Click on the Users link and then click on the Individual Users link.

Step 4 Click on the name of the user to select the toolbar to configure.

Step 5 Select *Enable Contact Center Integration*, as shown in Figure 11-16.

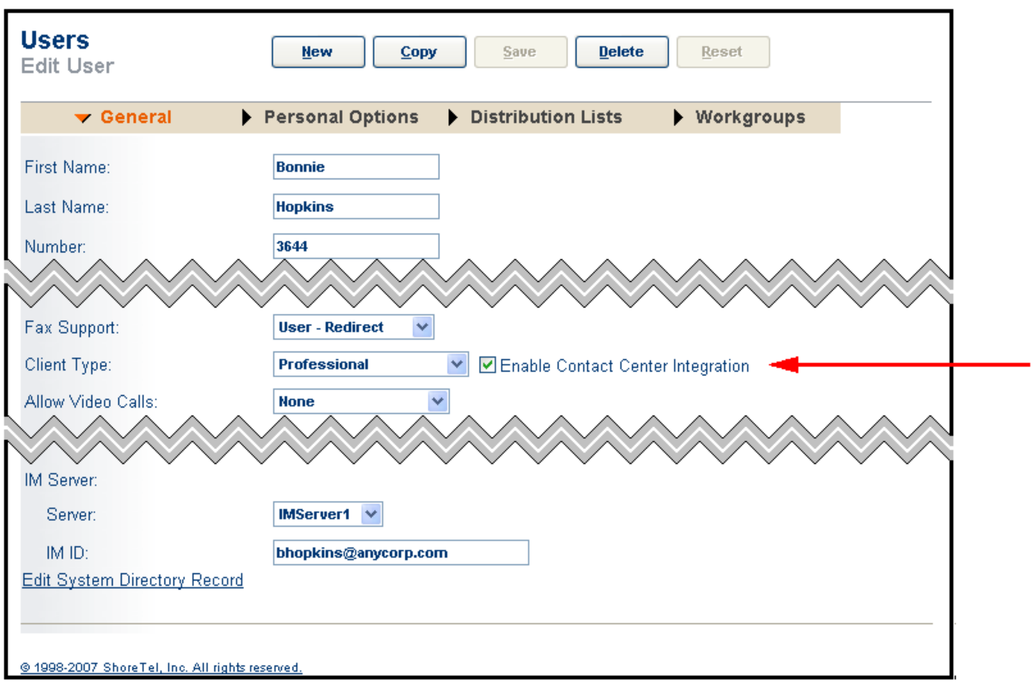


Figure 11-16 Selecting the Contact Center Client Type

Step 6 Click Save. The next task is to program the user's toolbar buttons.

To configure the programmable buttons toolbar for the selected user:

Step 1 Click the user's Personal Options tab.

Step 2 Click the Program Communicator Toolbars link near the center of the Personal Options window. If the user already has a Communicator toolbar configuration, a window like the example in Figure 11-17 opens. If no configuration exists, a blank space is displayed. For a new toolbar configuration, go to Step 3.

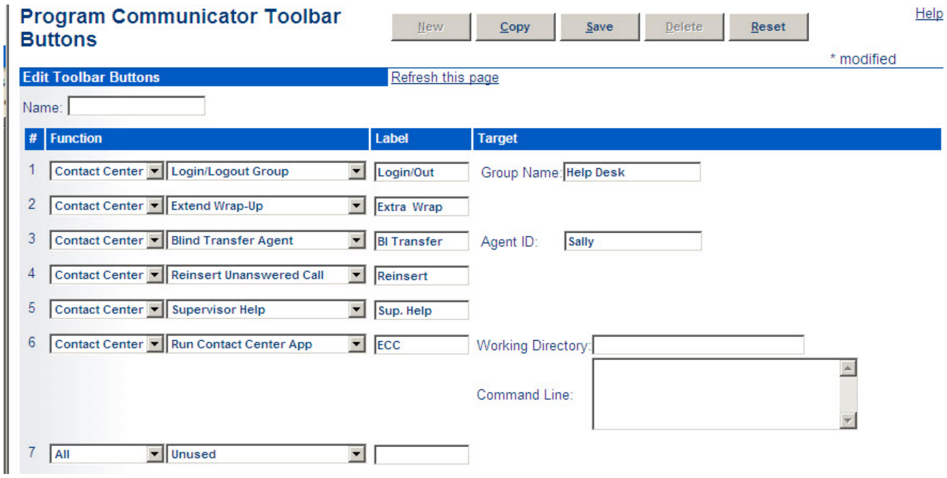


Figure 11-17 Assigning Contact Center Operations to Communicator Toolbar Buttons

Step 3 Click the New button to open a fresh button-programming screen.

The example in Figure 11-18 shows a new toolbar window in progress. It shows the choices and illustrates how the choice of operation can activate an additional parameter called Target. (Target can be a text entry box or drop-down menu.) In Figure 11-18, the two new function/operation combinations have triggered the appearance a Target option related to the selected operation. For the Contact Center's Blind Transfer Operation, the Target becomes the ID of the agent who should receive the transferred call.

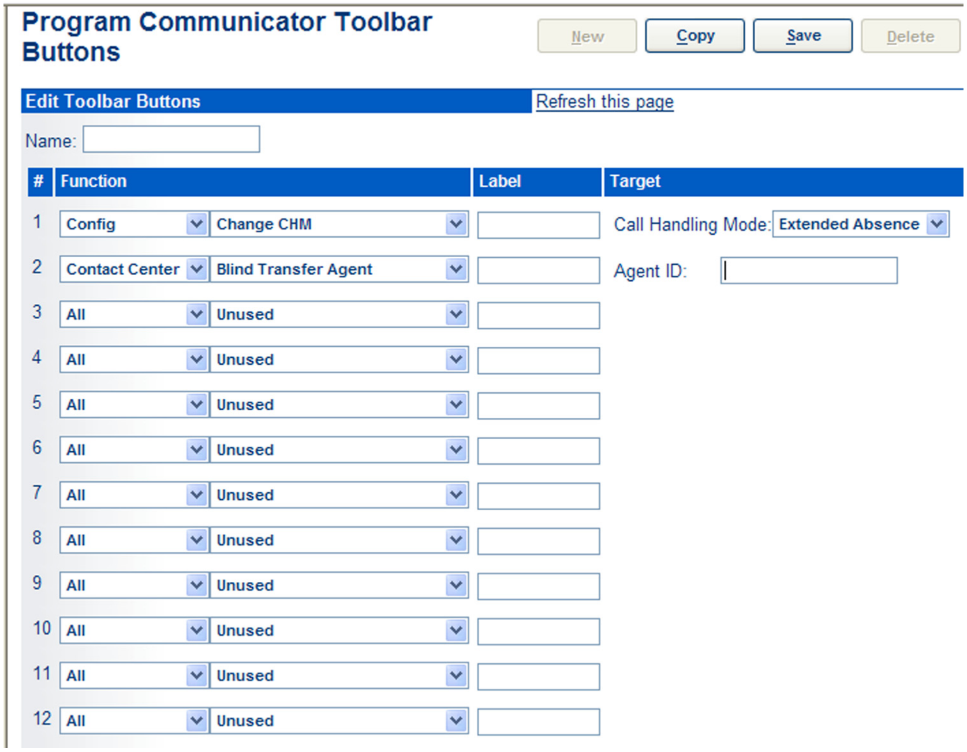
Step 4 Type a name for the new toolbar in the Name field.

Step 5 Select Contact Center from the drop-down menu under the Function heading.

Step 6 Select the operation from the drop-down menu to the right of the Function choice. This operation is associated with this toolbar button. (After the Contact Center function is chosen, the available operations apply to Contact Center only.)

Step 7 Type a label in Label field.

Figure 11-18 Configuring the Buttons of a New Toolbar



Step 8 Specify a target if the selected operation activates the Target option.

If a required target field for an operation is left blank or configured with invalid information, the system opens a dialog box to collect valid information when the user clicks on the associated button in Communicator. This pop-up dialogue box appears when inadequate information is configured for the following operations:

- Consult Transfer to Agent
- Blind Transfer to Agent
- Login/Login Group
- Logout Group
- Release with Reason Code
- Wrap-up with Code

Step 9 Continue programming the toolbar buttons as needed.

NOTE: If a button is unused or blank and this unused button sits between other, used buttons, Communicator interprets the sequence of blank buttons as a divider that is visible on the toolbar.

Step 10 When the toolbar button programming is done, click **Save**.

Repeat the process of assigning operations to buttons for each new user who is to use the integrated Communicator.

11.2.6 ECC Agent Queue

In a small call center or a sales-oriented call center, calls that are routed to an agent often must be queued for the agent when he or she is not available. The Agent Queue feature in Enterprise Contact Center allows calls to wait in a queue that is specific to an agent. Also, in conjunction with the Agent Queue feature, ShoreTel Communicator has been enhanced to let the agent manage his or her Enterprise Contact Center queue.

After a system administrator enables Transfer to Agent Queue for the individual agent-user in ShoreTel Director, the agent sees a Transfer to Agent Queue toolbar button in his or her ShoreTel Communicator. Subsequently, after logging into Enterprise Contact Center, the agent can press a button in Communicator to transfer an incoming call to his or her individual agent queue.

11.2.6.1 Implementation

This section describes how to enable the Agent Queue capability for an agent-user in Director. The Transfer to Agent Queue toolbar button is enabled in the same manner as other toolbar buttons in ShoreTel Director.

Refer to the Enterprise Contact Center Administrator Guide for additional information on the implementation of the feature.

To enable the Agent Queue capability in Director:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Users > Individual Users**. The Users page appears.

Step 3 Click the **Personal Options** tab.

Step 4 Click the **Program Communicator Toolbars** link. The Program Communicator Toolbars page appears.

Step 5 Click the **New** button or select the user you want to configure. The Program Communicator Toolbar Buttons page appears as shown in Figure .

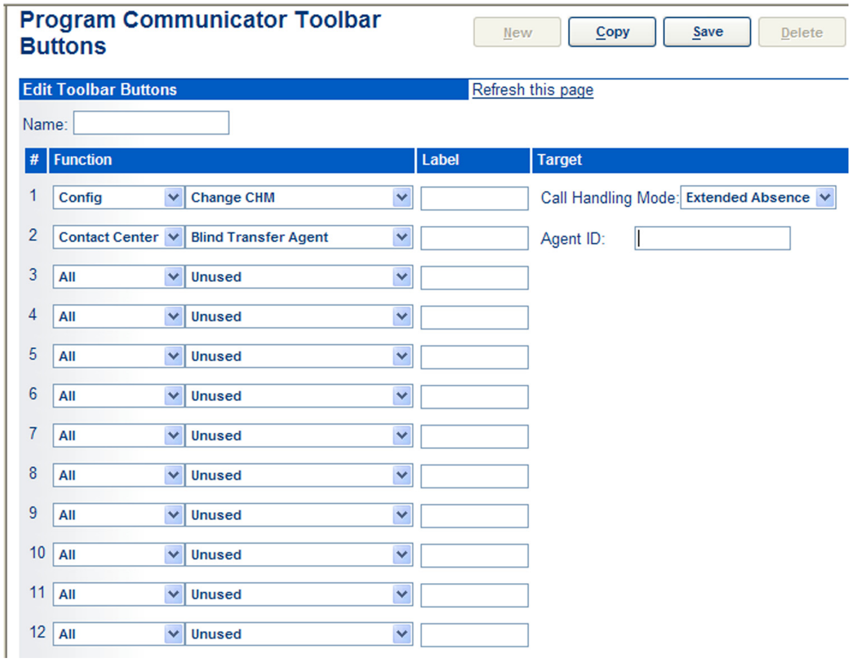


Figure 11-19Configuring Agent Queue Toolbar

Step 6 In the first Function field, select **Contact Center**.

Step 7 In the second Function field, select **Transfer to Agent Queue**.

Step 8 In the Label field, type the label for the button. You can use up to 12 characters for the label.

Step 9 Click **Save.**

11.2.6.2 Error Messages for ShoreTel Communicator Enhancements

ShoreTel Communicator lets an agent transfer an existing ACD call to an Agent Queue. In addition, the enhancements include failure messages for attempted transfers that fail and a warning message if the agent tries to log out while calls still exist in the Agent Queue.

Table describes the new messages that an agent might see.

11.2.7 ShoreTel Communicator Availability of Workgroup Controls to ECC Agents for Failover

Workgroup controls in the ShoreTel Communicator are available to Enterprise Contact Center (ECC) agents who use ShoreTel Communicator and are also members of a workgroup. The Login/Logout controls reflect their state in both ECC and the workgroups and are synchronized between these two systems.

Figure 11-20 shows the menu of ECC items for the Workgroup Agent Communicator when ECC integration has been enabled for the user. ECC control and status items are:

- “Logged Out of All Groups” (agent status)
- Set Agent ID

Table 11-2 Agent Queue Error Messages

Message	Description
You have queued calls in your personal queue. Press OK to logout of all queues or Cancel to remain logged in to just your personal queue.	When an agent logs out of all Enterprise Call Center groups from ShoreTel Communicator but still has calls queued in the Agent Queue, a warning message window pops up.
FAILMSG_FAILED_TO_SEND_CALL_TO_AGENT_QUEUE = 38	An attempt to transfer the call to the agent queue has failed.
FAILMSG_FAILED_TO_SEND_CALL_TO_AGENT_QUEUE_ALREADY_ANSWERED = 39	An attempted call transfer to an agent queue failed because the call was answered by another agent.
FAILMSG_FAILED_TO_SEND_CALL_TO_AGENT_QUEUE_NON_ACD = 40	An attempt to transfer the call to the agent queue has failed because the call is a non-ACD call.
FAILMSG_LOGOUT_FROM_AGENT_QUEUE_FAILED = 41	An attempt was made to logout all, logout primary, or logout specific from an agent queue while are calls were queued in the agent queue.

- Log In/Out
- Release/Resume
- Wrap-up Code
- Release with Code

Figure 11-20 Workgroup Agent Communicator with ECC Integration Enabled

11.2.7.1 Agent State Changes in ShoreTel Communicator

When an agent starts up ShoreTel Communicator, the agent's initial state is logged out of all ECC groups and all workgroups. Upon selecting Log Into All Groups, the agent logs into all ECC groups and all workgroups for which he or she is programmed (see Figure 11-21). Upon logging out, the agent is logged out of all ECC groups and all workgroups. If the ECC agent state changes to the Release state, the Workgroup state changes to the Wrap-Up state.

Note: We strongly recommend that an ECC agent be allowed to log into only workgroups that serve as a backup for the main ECC route. Receiving calls concurrently from ECC and Workgroups that are not backup workgroups might show inconsistencies in reporting of Non-ACD (NACD) calls in ECC.

Figure 11-21 Changing States by Using ShoreTel Communicator

11.2.7.2 Failover Behavior of ShoreTel Communicator

If ECC is unreachable or unavailable, ShoreTel Communicator indicates the system is in "Failover Mode" (while the agent is a member of a Workgroup), as shown in Figure 11-22. Failover Mode lets an agent continue to log in and out of workgroups through the Contact Center menu or the toolbar menu. As Figure 11-22 suggests, only the Log Into All Groups and Log Out of All Groups options remain available. Other options are grayed out.

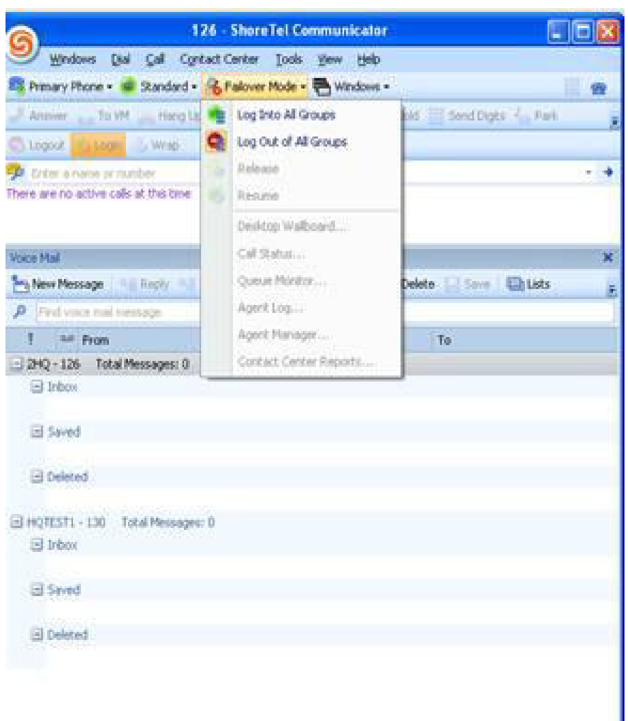


Figure 11-22 Failover Behavior of ShoreTel Communicator with Workgroup

If ECC is unreachable or unavailable and the agent is not a member of a Workgroup, ShoreTel Communicator displays "System Unavailable" as shown in Figure 11-23.

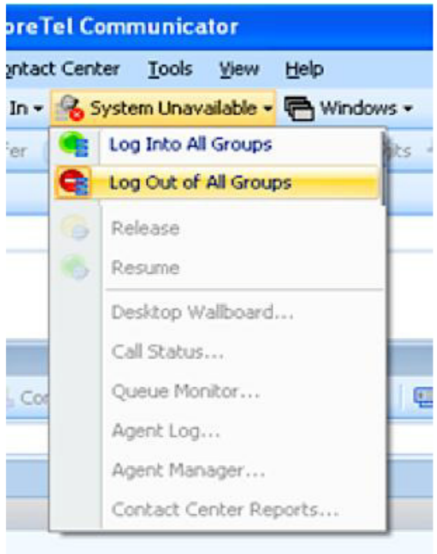


Figure 11-23 Failover Behavior of ShoreTel Communicator without Workgroup

11.2.7.3 Other Behaviors of ShoreTel Communicator

If ShoreTel Communicator loses contact with an ECC server, it keeps the login state locally. (The local state is the state the client was in before disconnection.) When connectivity resumes, ShoreTel Communicator logs in or logs out based on the local login state.

If an administrator disables Contact Center Integration in ShoreTel Director, ShoreTel Communicator logs out of all ECC groups but remains in the last known state that was set for the workgroups (any workgroup) of which the ShoreTel user is a member.

11.2.7.4 Workgroup Agent Contact Center Integration

To enable Contact Center Integration if you want the agent to use the Workgroup Agent Access level licensing as well follow these steps:

Step 1 Log into ShoreTel Director.

Step 2 Click on the **Users -> Individual Users** link (see Figure 11-24).

Step 3 In the Access License drop-down list, select Workgroup Agent.

Step 4 Put a check the Enable Contact Center Integration checkbox.

Step 5 Click on the **Save** button.

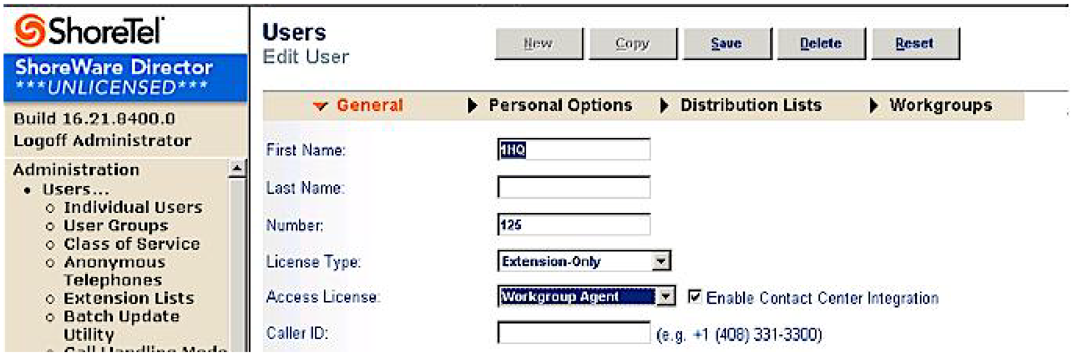


Figure 11-24 Selecting the Workgroup Agent Access License Configuration

11.3 Communicator for Mobile

Communicator for Mobile expands the remote user feature set by adding Communicator functionality to mobile devices.

Communicator for Mobile comprises the following components:

- **Communicator for Mobile Server (MCMS)** is a ShoreTel server component that manages all communications with ShoreTel Communicator for Mobile clients. Installing the ShoreTel Server includes the automatic installation of MCMS.
- The **Communicator for Mobile (CM)** client application is installed on each mobile device. Communicator for Mobile accesses ShoreTel functions, configuration information, voicemail, and calling history by communicating with the MCMS.

The MCMS is located on the main and distributed ShoreTel servers. Secure wireless communication sessions with specified mobile devices are conducted through a Reverse Proxy Server or a Blackberry Enterprise Server.

Mobile devices receive MCM Client Installation Files from the ShoreTel Server through a Blackberry Enterprise Server. After installing these files, the end user can configure the mobile device to access the ShoreTel server through an IP address provided by the system administrator.

ShoreTel Communicator for Mobile enables a mobile device to function as a ShoreTel extension and provides an interface, similar to Communicator, for accessing ShoreTel client information. In addition to initiating calls from the Call History, Voice Mail, and QuickDialer screens, end users can access their voice mail, configure call handling mode settings, set the active call handling mode, configure Extension Assignment settings, and activate Extension Assignment from the ShoreTel Communicator for Mobile user interface.

Communicator for Mobile supports up to 200 conventional users per ShoreTel system.

11.3.1 Supported Configurations

11.3.1.1 ShoreTel Communicator for Mobile Support – Languages

Communicator for Mobile software supports all elements of English (U.S.), English (U.K.), German, Spanish, Swedish, French, Dutch, Danish, Italian, and Norwegian.

Usability Enhancements

Communicator for Mobile now supports ***smart type***. Smart type means that certain keys do not need a special key to enable character entry. On a BlackBerry, for example, a user can type a "." or "-" without using the Alt key on numeric fields. This addition expedites the entry of IP addresses or extension numbers.

The left and right direction keys move the cursor so the user can insert one or more characters instead of being required to delete an entire field to correct an error. This feature is available on most supported devices.

Localization Features

Localization support affects the following area of the product:

- **Western languages:** the supported languages are English (U.S.), English (U.K.), German, Spanish, Swedish, French, Dutch, Danish, Italian, and Norwegian.
- **Keyboard types:** on devices that have full keyboards, users can choose from programmed keyboard layouts for their location. For example, the top row of letters on American keyboards includes q w e r t y; German keyboards have q w e r t z; and French keyboards have a z e r t y.
- **Character set:** Communicator for Mobile utilizes the Latin-1 character set (see http://htmlhelp.com/reference/charset/). This character set includes all accented characters and other global symbols.
- **Ordering of time and date:** based on the language that a user specifies, Communicator for Mobile orders the date and time data according to the custom for that language. For example, with U.S. English, the order is mm/dd/yy; for U.K. English, dd/mm/yy; and for Dutch, dd-mm-yy. A table with the format of dates and times appears in the "Localized Date and Time Table" on page 405.
- **Accented characters:** Communicator for Mobile supports accented characters (diacritical marks). The keyboard for the selected language supports these marks, and the QuickDialer key filter also supports diacritical marks.
- The thumb wheel on the side of the device or other rotating character facility lets the user enter accented characters on the phone. The specific method for rotating through a character set depends on the model, yet the implantation is specific to Communicator for Mobile. (The ShoreTel model mirrors how a user enters characters on the device.)
- **Localized time on the phone:** The server uses a standard GMT format that the client converts to the time according to the time zone that is detected by the device. Therefore, the phone shows the time according to the time zone where the phone is located.

11.3.2 ShoreTel Communicator for Mobile Support – Devices

ShoreTel Communicator for Mobile is supported on the following devices:

- Samsung BlackJackII (New in ShoreTel 11)
- BlackBerry 81xx Series
- BlackBerry 83xx Series
- BlackBerry 88xx Series
- BlackBerry 90xx Series

- BlackBerry 8900
- BlackBerry 9500
- Nokia E61i
- Nokia E65
- Nokia E71
- Nokia E72
- Nokia E75
- Nokia Surge 6790
- Nokia E90
- Nokia N78
- Nokia N82
- Nokia N95
- HTC Mogul (Sprint PPC-6800)
- HTC P6500 (Sirius)
- HTC TyTn II

ShoreTel Communicator for iPhone

- iPhone OS 3.1 and later
- iPhone 3G and 3GS
- iPhone 4

The following section describes a few functional differences based on the manufacturer or by model. Most models support all functions in Communicator for Mobile. This section lists the exceptions, so every model supports every function of Communicator for Mobile unless noted under a heading in this section.

11.3.2.1 Carrier Support

ShoreTel Communicator for Mobile is supported across carriers that allow installation of third-party applications onto the BlackBerry device.

The BlackBerry series 8100, 8300, 8700, and 8800 and the BlackBerry 7290 models are supported on any network that allows the installation of third party software. Although some mobile carriers restrict the downloading and installation of third-party software through their network, no known restrictions for the BlackBerry devices currently exist in any network.

Carrier support and restrictions on Nokia E65 series devices can be addressed by the end user's carrier.

ShoreTel Communicator for Mobile supports LBS on the Nokia E71.

Devices supporting LBS must either have internal GPS hardware or the capability of connecting to an external device that provides GPS functionality, such as dongles. The carrier must allow GPS on the device and allow third party applications, such as MCM, to use GPS functionality. External GPS devices may be used to enable LBS features on these devices, depending on carrier limitations.

11.3.2.2 PIM Integration

The Nokia E71 supports PIM entry import. The ShoreTel Communicator for Mobile settings page provides an option to disable PIM loading. The TY TN II does not support PIM.

11.3.2.3 Screen Transition

Screen transition refers to leaving one screen and going to another screen. Manufacturers offer menu-operations, buttons, or softkeys for moving to another screen. This section lists restrictions that manufacturers have for screen transitions by the clicking of a menu item, physical button, or soft key.

- Menu: The Nokia E65 does not have menu-activated screen transitions.
- E65 screen transitions are done by softkeys.
- Button: None of the current models have button-only screen change activation.
- Soft key: Soft key is not supported by any of the BlackBerrys.

11.3.2.4 Personal Information Manager

Personal Information Manager (PIM) lets a device synchronize information from the Quickdialer feature in Personal Communicator so that this information is automatically loaded and updated in Communicator for Mobile. PIM is supported by all supported devices.

11.3.2.5 Preview Voicemail

Voicemail cannot be previewed on the BB7290 because this model has no external speaker.

11.3.2.6 Font Size

Font size in the display cannot be changed on any of the BlackBerry models.

11.3.3 Requirements

11.3.3.1 MCMS

The Communicator for Mobile Server (MCMS) is the ShoreTel server component that manages ShoreTel Communicator for Mobile client communications. The MCMS is installed on the ShoreTel HQ and DVM servers as part of the normal ShoreTel Server installation process. The MCMS contains no configuration parameters and requires no administrator intervention or monitoring during normal ShoreTel operations.

11.3.3.2 MCM Client Installation Files

Communicator for Mobile Client Installation Files are a set of files that provide the ShoreTel Communicator for Mobile client application to mobile devices. Clients download these files to their mobile devices from a server specified by the system administrator. Communicator for Mobile Client Installation Files are maintained and updated by the system administrator.

These files are originally placed on the ShoreTel Server as part of the ShoreTel installation, then moved by the system administrator to a server that can be accessed by the mobile devices of system end users.

11.3.3.3 Servers

Mobile devices must connect with the ShoreTel server to access MCMS services. Enterprise security requirements affect the structure of the network that provides application support. The following section describes servers that are typically used to implement ShoreTel Communicator for Mobile.

11.3.3.4 Blackberry Enterprise Server (BES)

The Blackberry Enterprise Server is a middleware application, offered by Research in Motion, that supports ShoreTel Communicator for Mobile by synchronizing email and PIM information through a secure back channel between the MCMS and BlackBerry mobile devices. The BES also supports Client Installation File downloads from ShoreTel to BlackBerry devices.

11.3.3.5 Reverse Proxy Server

A reverse proxy server is a proxy server that is normally installed in front of web servers. Internet communications addressed to a web server are routed through the reverse proxy server, which can process the request or pass the request to the specified web server. Implementing a reverse proxy server within the corporate DMZ maintains the integrity of the corporate firewall and insulates the ShoreTel Server, along with other corporate assets, from public networks.

A reverse proxy server provides secure wireless communication between Mobile Devices running ShoreTel Communicator for Mobile and ShoreTel server. A reverse proxy server can serve the same role as a BES for conducting ShoreTel Communicator for Mobile wireless communication sessions.

To fully set up a reverse proxy you would need an Apache server version 2.2 or higher, and a SSL certificate from a root certificate authority. The system supports a self-signed certificate, however, the users will receive a warning each time the application is launched. This is not recommended for production deployments

See the ***ShoreTel 12: Planning and Installation Guide*** for details on configuring reverse proxy services.

Direct communication between an MCMS with the mobile devices is not recommended. This configuration compromises the integrity of the corporate firewall and exposes the ShoreTel Server and other connected corporate assets to attacks from public networks.

11.3.4 Administrator Configuration

ShoreTel Communicator for Mobile administration comprises two application activities: client application installation file delivery to the mobile devices and ShoreTel Communicator for Mobile communication sessions with the mobile devices. Configuring ShoreTel to support ShoreTel Communicator for Mobile requires the completion of the following tasks:

- Installing at least one additional server
- Acquiring the required ShoreTel Communicator for Mobile license
- Authorizing clients to use ShoreTel Communicator for Mobile

MCMS installation is performed with the installation of a ShoreTel server. The MCMS requires no additional installation or configuration. The following sections describe the tasks required to install and support ShoreTel Communicator for Mobile.

11.3.4.1 Server Installation (Headquarters and DVM)

Server configuration options:

- ShoreTel Communicator for Mobile communication sessions with a BlackBerry device are supported by a BES or a Reverse Proxy Server.
- Client Installation File downloads to a BlackBerry are supported through a BES.

Systems can use a BES server for all ShoreTel Communicator for Mobile activities.

BlackBerry Enterprise Server (BES)

The BlackBerry Enterprise Server supports ShoreTel Communicator for Mobile communications between the ShoreTel server and BlackBerry devices. BlackBerry devices can also access MCM Client Installation Files located on the ShoreTel server through the BES. The BES implements a secure connection, similar to that of a VPN, to provide access for mobile devices to resources that are protected by the corporate firewall.

For information on installing a BES, refer to BES installation instructions provided by Research in Motion.

11.3.4.2 Client Installation Files

Installing the ShoreTel Server places the Communicator for Mobile Client Installation Files directly on the ShoreTel Server. BlackBerry devices can access the files directly from the ShoreTel Server if the network includes a BES.

Version numbering for Communicator for Mobile is in sync with the ShoreTel version number. The download format is xx.yy.zz. This format is required by the Nokia E65. In contrast, the BlackBerry does not have such a requirement. Nevertheless, an image with a number like 13.20.8700.0 is listed on the server as 13.20.87.00 for download activities, yet it appears in the device's About window as 13.20.8700.0.

Client Installation files can be delivered from a system that runs Windows 2003 or 2008 Server and has IIS. The IIS server must be configured to serve WML (Wireless Markup Language) files to the mobile devices. Table 11-3 lists the MIME types required to serve the installation files.

Table 11-3 Mime Types Required to Serve MCM Installation Files

File Extension	MIME Type
.jad	text/vnd.sun.j2me.app-descriptor
.jar	application/java-archive
.wml	text/vnd.wap.wml
.cod	application/vnd.rim.cod

To register the required MIME types:

Step 1 On the computer desktop, click **Start > Settings > Control Panel**. The Control Panel appears.

Step 2 Select **Administrative Tools**.

Step 3 Select **Internet Information Service**.

Step 4 Open the HTTP Headers page, shown in Figure 11-25, by clicking the **HTTP Headers** tab at the top of the Internet Information Services page.

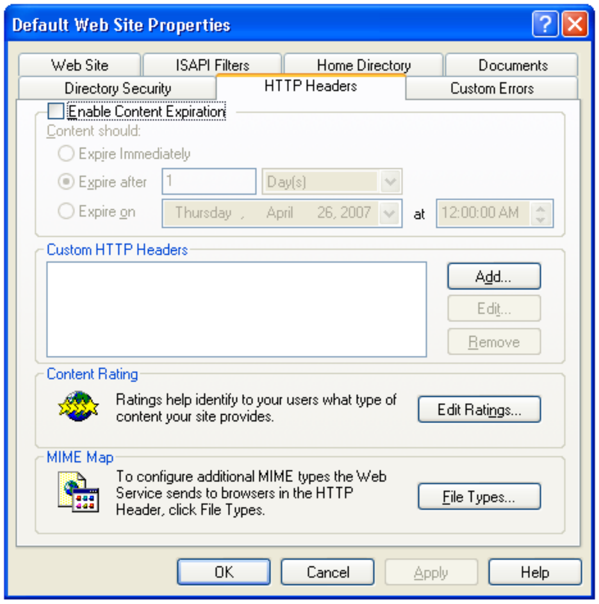


Figure 11-25Internet Information Services Page – HTTP Headers Page

Step 5 Open the MIME Types page by clicking the **MIME Types...** button in the bottom right corner of the HTTP Headers page.

Step 6 Open the MIME Types Dialog box by clicking the **New** button on the top right corner of the MIME Types page, as shown in Figure 11-26.

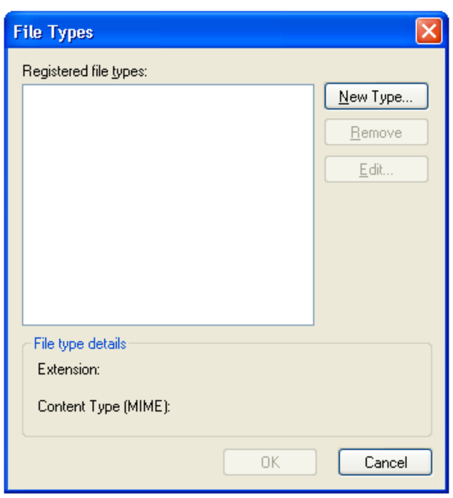


Figure 11-26MIME Types Page

Step 7 Enter the Associated extension and content type data in the corresponding data entry field of the MIME Types Dialog box, shown in Figure 11-27, for one of the following types:

- Content type: **text/vnd.sun.j2me.app-descriptor**; extension: **.jad**
- Content type: **application/java-archive**; extension: **.jar**
- Content type: **text/vnd.wap.wml**; extension: **.wml**
- Content type: **application/vnd.rim.cod**; extension: **.cod**

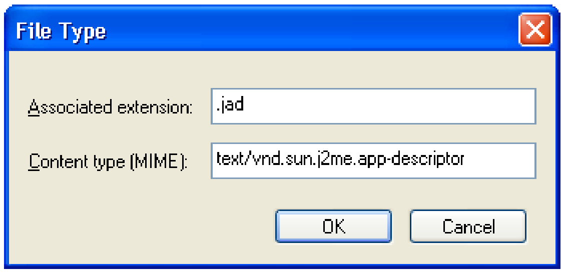


Figure 11-27MIME Types Dialog Box

Step 8 Click the **OK** button to close the MIME Types dialog box and return to the MIMES type page.

The table lists the type entered in Step 7.

Step 9 Repeat Step 6 through Step 8 for each type listed in Step 7.

Step 10 Click **OK** buttons on each successive page to save your changes and return to the desktop.

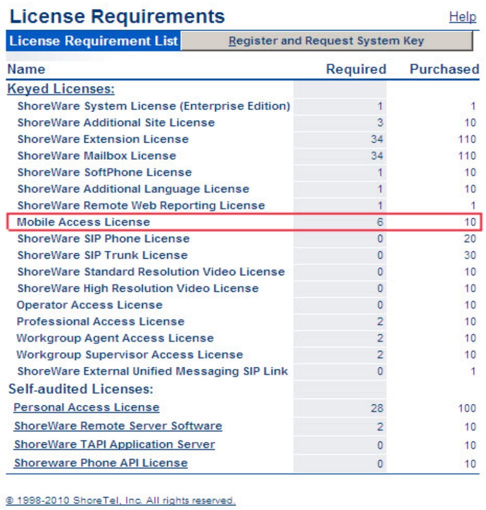
11.3.5 ShoreTel Director Configuration

11.3.5.1 Licenses

One ShoreTel Communicator for Mobile Keyed License is required for each client that is enabled for ShoreTel Communicator for Mobile. To view the number of ShoreTel Communicator for Mobile licenses available on the system, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration** > **System Parameters** > **Licenses** > **Requirements**. The License Requirements List page appears as shown in Figure 11-28.



License Requirements — Help

License Requirement List | Register and Request System Key

Name	Required	Purchased
Keyed Licenses:		
ShoreWare System License (Enterprise Edition)	1	1
ShoreWare Additional Site License	3	10
ShoreWare Extension License	34	110
ShoreWare Mailbox License	34	110
ShoreWare SoftPhone License	1	10
ShoreWare Additional Language License	1	10
ShoreWare Remote Web Reporting License	1	1
Mobile Access License	6	10
ShoreWare SIP Phone License	0	20
ShoreWare SIP Trunk License	0	30
ShoreWare Standard Resolution Video License	0	10
ShoreWare High Resolution Video License	0	10
Operator Access License	0	10
Professional Access License	2	10
Workgroup Agent Access License	2	10
Workgroup Supervisor Access License	2	10
ShoreWare External Unified Messaging SIP Link	0	1
Self-audited Licenses:		
Personal Access License	28	100
ShoreWare Remote Server Software	2	10
ShoreWare TAPI Application Server	0	10
Shoreware Phone API License	0	10

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Figure 11-28License Requirements Page

Step 3 In the Keyed License section, locate the Mobile Access License listing and check the Configured and Purchased columns for information about your mobile license status.

11.3.5.2 Client Authorization

ShoreTel administrators grant ShoreTel Communicator for Mobile access to clients through ShoreTel Director. To enable ShoreTel Communicator for Mobile access for a ShoreTel user:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Users > Individual Users**.

Step 3 Click the name of the user in the User List. The Edit User page appears as shown in Figure 11-29.

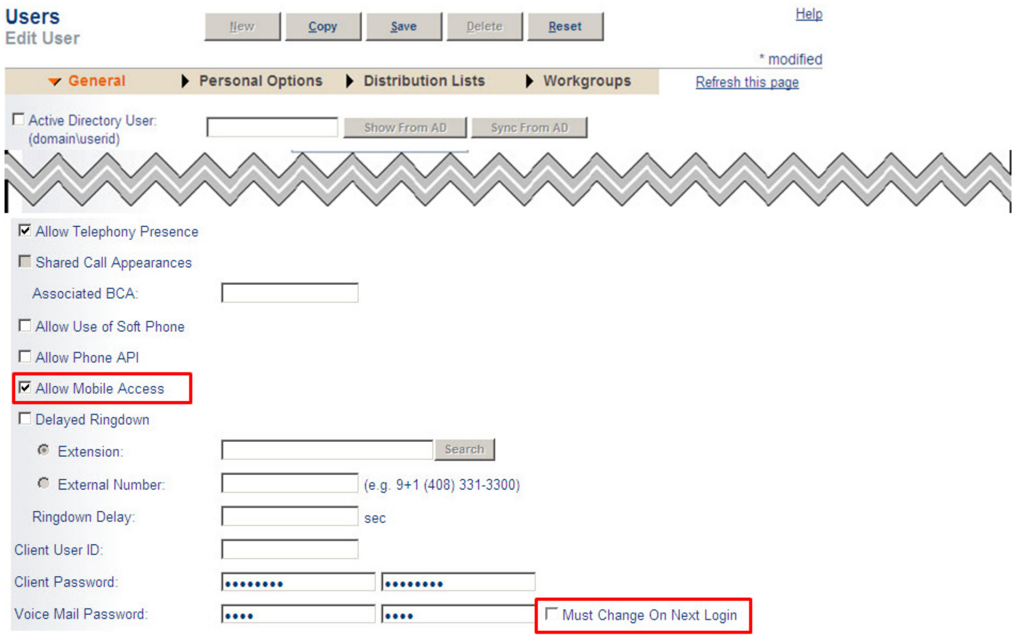


Figure 11-29Configuring an End User for ShoreTel Communicator for Mobile

Step 4 Check the **Allow Mobile Access** check box.

Step 5 Locate the Voice Mail Password fields and uncheck the **Must Change on Next Login** check.

Step 6 Click **Save**.

11.3.6 Client Configuration Information Delivery

The administrator must provide the following information to each client authorized to use ShoreTel Communicator for Mobile:

- URL of the server through which mobile devices download the Client Installation Files.
- IP address and Port Number through which the mobile device accesses the MCMS.
- the client's user extension.
- the client's voicemail password.

11.3.7 Configuring Information upon First-Time Use

This section describes the first-time configuration requirements of the device.

11.3.7.1 Languages

The first screen that a user sees upon first-time startup of Communicator for Mobile under ShoreTel 12 is the language selection screen (Figure 11-30). The supported languages are English (U.S.), English (U.K.), German, Spanish, Swedish, French, Dutch, Danish, Italian, and Norwegian.

Figure 11-30Language Selection Screen

Step 1 To select a language, use the track ball, wheel (at the right side of the device), or other device-specific selector to highlight a language.

Step 2 Press the wheel or press Select to use the highlighted language.

On devices that have a full keyboard, the keyboard selection screen appears after language selection. For devices that do not have a full keyboard, users are taken to the provisioning window.

11.3.7.2 Keyboard Type

After selecting a language, users see the keyboard selection menu on devices that have a full keyboard as shown in Figure 11-31:

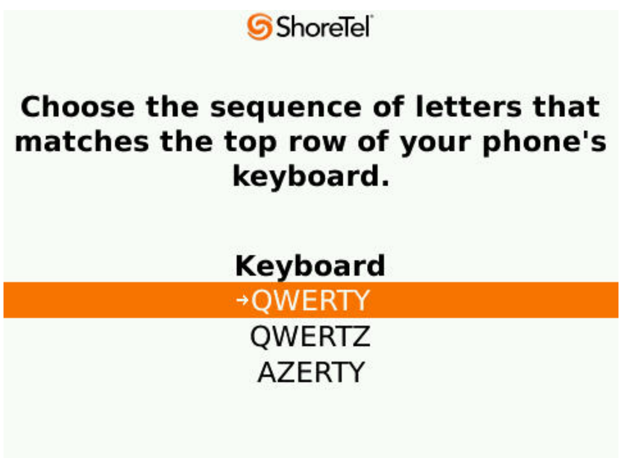


Figure 11-31Keyboard Selection

Step 1 To select a keyboard, use the track ball or wheel at the right side of the device (or other, device-specific selector) to highlight the sequence of letters on the top row of the keyboard.

Step 2 Press the wheel or press Select to use the highlighted keyboard. After the user selects a keyboard, the informational screen appears in the selected language.

Step 3 Press **Next** to open the next screen, where specific user-details are entered. See next section, Provisioning User Details.

11.3.7.3 Provisioning User Details

The following user information is required for the first-time provisioning of a new phone:

- The IP address of the server that hosts the user account (not the DNS) is obtained from a system administrator. This server can be the BES or some other mechanism for reaching the server from the outside, such as a reverse proxy server.
- The Port number on the server that hosts the user account (obtained a system administrator).
- The user's extension.
- The password is the personal ID number (PIN): the user must enter this ID before hearing voicemail. Upon power up after the first-time specification of the password, the User Information screen does not show the password field.

After these four values are entered upon the initial run, Communicator for Mobile retains the configuration so the user does not need to re-enter it. An example of the regular User Information screen that is displayed after the initial configuration appears in Figure 11-32.

Figure 11-32User Information Screen

The user can change the startup screen through the Settings window ("Default Start Page" in Figure 11-33).

Figure 11-33Settings Screen

After user information is initially entered, a status message momentarily appears, followed by a welcome message. After the welcome message, the Main Menu appears as the default startup window. The user can choose a different startup window by using the Settings window.

11.3.7.4 Specifying Accented Characters

The rotating character facility lets the user enter accented characters on the phone. This scheme follows the model used on the manufacturer's device. However, the implantation is specific to Communicator for Mobile, and this model mirrors the way a user enters characters on the device.

If a user enters the string "Andre" but actually wants "André Dupont," a wide variety of possibilities can come up (if they exist in the user's directory), for example:

Andre, Andrè, André, Andrê, Andrë, AndrÈ, AndrÉ, AndrÊ, AndrË . . . .

Now, the user can enter "André" to get André Dupont. Quickdialer can use the specific device's scrolling mechanism to select an accented character. Different devices have different mechanisms for scrolling through character choices. For the BlackBerry models, the user holds down a key and uses the trackball or thumb-wheel to scroll through the possibilities. Furthermore, all accent marks are available regardless of the selected language.

11.3.7.5 Localized Date and Time Table

This section consists of a table (Table 11-4) that lists the date and time that appears on a device based on the selected language.

Table 11-4 Date and Time Formats by Language (Country)

Language	Date	Time
Deutsch	dd.mm.yy	08:48 23:59
Español	dd/mm/yy	8:48 23:59
Svenska	yy-mm-dd	08:48 23.59
Français	dd/mm/yy	08h48 23h59
Nederlands	dd-mm-yy	8:48 23:59
Dansk	dd-mm-yy	8:48 23:59
Italiano	dd/mm/yy	8:48 23:59

Table 11-4 Date and Time Formats by Language (Country)

Language	Date	Time
Norsk	dd.mm.yy	08h48 23h59
English U.K.	dd/mm/yy	08h48 23h59
English U.S.	mm/dd/yy	8:48 AM 11:59 PM

11.3.8 Upgrading Communicator for Mobile

For an administrator, upgrading Communicator for Mobile requires only the installation of ShoreTel Version 12 to the server that runs Communicator for Mobile. This installation automatically places new versions of Communicator for Mobile's server components and client installer on the server.

Before upgrading a client device, the subscriber must delete the existing application from the device. To perform the upgrade, the subscriber accesses the URL that was used to download client software. Access to the Communicator for Mobile is available through the following URL format:

http://*server*/mcm/client/

where *server* is either a valid DNS name or IP address of the Communicator for Mobile server. Upon entry of the correct URL, the device's browser is automatically directed to the latest file.

The server needs to be reachable from the mobile phone. This server could be a BlackBerry Enterprise Server (BES) or some other mechanism for reaching the server from the outside, such as a reverse proxy.

The following sequence illustrates an installation from a BES server:

1. The system administrator sends an email with a link to the location of the Communicator for Mobile software to the subscriber.
2. On the mobile device, the subscriber clicks on the email's enclosed link to get access to the download site.
3. The subscriber removes the current MCM version from the device.
4. The subscriber clicks the link leading to the download site on the browser window.
5. After the Communicator for Mobile software has been downloaded, the subscriber configures the client with the IP address and port allocated on the ShoreTel server for Communicator for Mobile.

As long as the device can reach the ShoreTel server, no other configuration or setup is required between the BES and the ShoreTel Server. (A company could restrict on access to its BES.)

Alternatively, the user can open a browser on the device and enter the URL of the server (in the format http://*server*/mcm/client). The server window opens with choices for locating the newest client software, as Figure 11-34 shows.

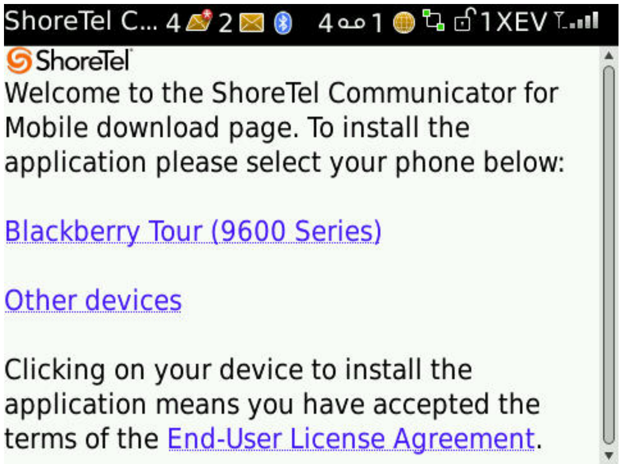


Figure 11-34Accessing the Newest Client Software

The user selects the application for the device. In Figure 11-35, the software is for a BlackBerry 8800 series.

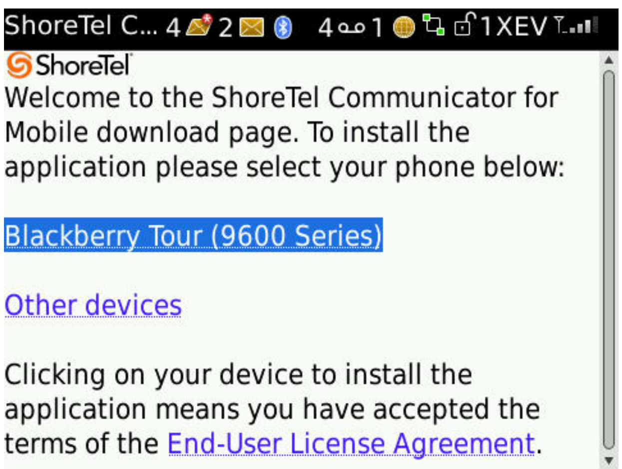


Figure 11-35Choosing Specific Client Software

To download the client software, the user highlights and presses the Download button (Figure 11-36).

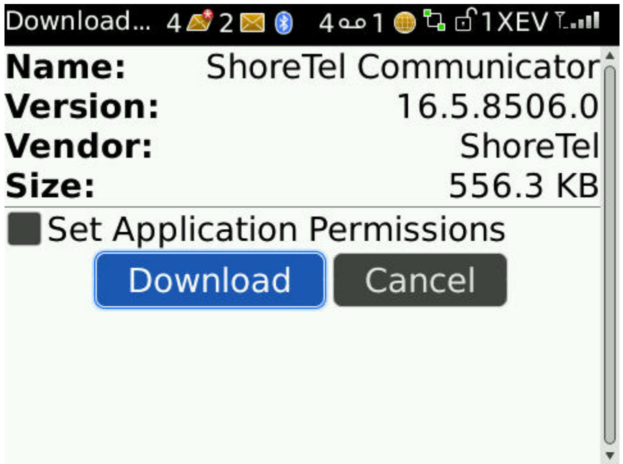


Figure 11-36Starting the Download of the Selected Client Software

Upon successful software download a message (Figure 11-37) shows that the device is ready for the user to make choices for Communicator for Mobile's operation.

Figure 11-37After Successful Download

CHAPTER 12

Configuring User Features

This chapter is about setting up users in the ShoreTel system. The topics discussed include:

12.1 Private Numbers

Users can have private numbers that are not listed in the System Directory or in Communicator Quick Dialer and thus have Caller ID information suppressed. This is enabled through the check box on the Edit User page called Make Number Private. When checked, the user's extension becomes a Private Number.

The following conditions apply to private numbers:

- The user's Private Number extension no longer appears in the QuickDialer for dial-by-name operations or in the ShoreTel Directory Viewer.
- Calls placed by a Private Number user show the caller's name but not their number to the dialed party for internal calls. This applies to analog phones, IP phones, and associated instances of the Communicator. The ring style is a double-ring, indicating an internal call.
- Calls placed from a private number to an external party do not deliver a Direct-Inward-Dial (DID) number as Caller ID when PRI trunks are used for the outbound call. The proper CESID (Caller's Emergency Service ID) will only be delivered for 911 calls.
- Calls placed from a private number to an off-system extension on PRI trunks with NI2 signaling deliver calling name information but not calling number information.
- Routing slips and the Communicator and History viewer show the Private Number user's name but not their extension number.
- The Private Number users are listed with name and number in the Extension Monitor extension selection dialog box.
- The user can be dialed directly via the telephone or the Communicator if their extension is known.
- Contacts imported from Microsoft Outlook or Exchange that reference a Private Number user's extension are not blocked and are fully visible in the Communicator Quick Dialer.

- CDR database records show both number and name for Private Number users. The Caller-ID Flags field will indicate that only the name is valid however.
- CDR legacy log files show the number of Private Number user calls that are inbound or outbound calls.
- The ShoreTel Director shows number information for Private Number users as with other users, as for example on the User List page.

12.2 Using Extension Assignment

Extension Assignment is a feature of the ShoreTel system that allows a user to quickly and easily re-assign his or her extension to any telephone on the system or off the system. The user's communications profile is reassigned to that telephone, and calls placed to the user will be routed to that telephone, while calls placed from that telephone will reflect proper caller ID information.

The Extension Assignment feature might be used by the following types of users:

- Multi-site users, such as executives or managers, who might use the system across multiple locations. Extension Assignment allows these users to pick up a telephone at any location on the enterprise network, log into voice mail, and assign their extension to that telephone. (See the "Configuring On-Net Extension Assignment" on page 417.)
- Office hotel users, such as contractors or telecommuters, who may occasionally be out of the office or who might share a cubicle (and thus a phone) with another employee. Extension Assignment allows these users to have their own extension and mailbox, yet not have a dedicated switch port. They can simply assign their extension to a telephone on the network when they are in the office while allowing another user to usurp that phone when they are done with it. (See the "Configuring On-Net Extension Assignment" on page 417.)
- Remote or mobile users, such as employees in sales or support, who might travel frequently and would like to have all calls directed to their cell phone or home office PSTN phone. Extension Assignment allows these users to have their own extension and mailbox, yet not have a dedicated switch port, thus optimizing system resources. (See the "Configuring Off-Net External Assignment" on page 414.)
- Legacy PBX Users, such as users with Off-System Extensions working with Communicator. (See the "Configuring Off-Net External Assignment" on page 414.)

Extension Assignment also allows the system administrator to configure all telephones as anonymous telephones and all users as virtual users, eliminating administrative costs associated with frequent moves. When a move occurs, users simply assign their extension to the telephone at the new location.

The feature also allows an off network user to manage a call via Communicator, so while the conversation occurs over the cell phone or home phone, the call appears via Communicator and can be controlled using many of the features available via Communicator. Note that this requires the user to be located near a PC that has access to a broadband connection and that Communicator is running.

Other benefits to the (Off-Net) Extension Assignment user include:

- Use the existing PSTN line for voice while managing the call via Communicator over an ordinary broadband Internet connection.

- Emulate analog extension hook switch actions via star-star (**) for FLASH and pound-pound (##) for on-hook/off-hook.
- Access the user's directory numbers at the office.
- The user appears to be calling from the office.
- Keep communication costs minimized with flexible IP & trunking requirements.
- Retain call management features of the ShoreTel system over a broadband connection while maintaining audio quality over PSTN.

12.2.1 Special Considerations

The following list shows considerations for Extension Assignment:

- When an Extension Assignment call finishes but the carrier has not reported back to the ShoreTel system that the far side has disconnected, the call remains active for approximately five seconds before it is finally disconnected. To initiate a new call during the five-second window, the user can press "##."
- To use the Extension Assignment feature, the user must be in a user group that has a Telephony Class of Service with ***Allow Extension Reassignment***. See "Configuring Permission Settings" on page 412.
- SIP is not supported as the external number to which the user is reassigned.
- ShoreTel supports up to 1,000 virtual users.
- Incoming calls to a user's extension that has been assigned to an off-net location rings on the cell phone or Off-System Extension. If the call is not answered, normal call handling allows the caller to leave a message in the user's ShoreTel mailbox.
- Extension Assignment, when assigned to an off-net location, is fully controllable through ShoreTel Communicator (except for answering a call, which must be done manually). Also, Extension Assignment has limited TUI functionality.
- Extension Assignment calls that are terminated through ShoreTel Communicator are not followed by the standard dial tone. Extension Assignment uses a unique set of internal and external dial tones. This difference in tones can be important in installations where network devices have been configured to listen for normal class progress tones before taking action on a call, such as hanging up.
- Calls placed or answered through Extension Assignment, when assigned to an off-net location, continue to show in the ShoreTel Communicator call stack. Normal call control functions, such as hold, unhold, conference, transfer, and park, continue to work. Park to the Extension Assignment extension, when assigned to an off-net location, is not supported.
- Extension Assignment, when assigned to an off-net location, behaves like an automated "Find Me" feature except that the caller does not press 1 to find the called party. (The PSTN phone number is immediately called.) The call recipient can answer the call by lifting the handset (or activating a cell phone) and pressing the DTMF digit 1 in response to the repeating prompt.
- Prompts (such as those used for Find Me) can be used to confirm answering. The answer style can be configured to be one of the following possibilities:
  - — Wait for DTMF (the default) - The call is not forwarded until the user presses DTMF 1.

— Wait for Answer - The ShoreTel system forwards the call as soon it detects the far-end answer. (Note that far-end answer detection is not supported by the central office for some trunk types, in which case the user must press 1.) (DTMF is audible to the caller.)

12.2.2 Terminology

The terms used to describe Extension Assignment are as follows:

- **Anonymous telephone:** A telephone not currently assigned a user. You can make a call from an anonymous telephone, but you cannot call an anonymous telephone.
- **Any IP Phone:** The feature that lets users assign their extension to any IP phone on the enterprise network.
- **Assign:** The command that assigns an extension to a telephone.
- **Assigned:** The status of a user who is currently assigned to a telephone that is not their home phone.
- **Current telephone:** The telephone to which the user is currently assigned (also known as the current switch port).
- **Go Home:** The command to assign a user's extension back to his or her home telephone.
- **Home:** The status of a user who is assigned to his or her home telephone.
- **Home telephone:** The telephone to which the user is normally assigned (also know as the home switch port). This is the telephone to which the user returns when using the Go Home command.
- **Extension Assignment:** The feature that lets users assign their extension to any telephone, on-system or off-system extension.
- **Unassign:** The command that unassigns an extension from a telephone.
- **Vacated telephone:** A home telephone that currently does not have a user assigned. These are listed on the Anonymous Telephones edit page under Vacated Telephones.
- **Virtual user:** A user who does not have a physical telephone port and is currently assigned to the SoftSwitch.

12.2.3 Configuring Permission Settings

To configure the Extension Assignment feature, the system administrator must first give the end user permission to use the feature by following the procedure below:

Step 1 Launch **ShoreTel Director** and enter the user ID and password.

Step 2 Click on the **Administration** link to expand the list (if it has not already been expanded).

Step 3 Click on the **User Groups** link.

Step 4 Create a new user group for the Extension Assignment users, or click on the name of an existing user group (in which all members will have access to Extension Assignment).

Step 5 For the COS - Telephony field, click on the **Go to this Class of Service** link to display as shown in Figure 12-1.

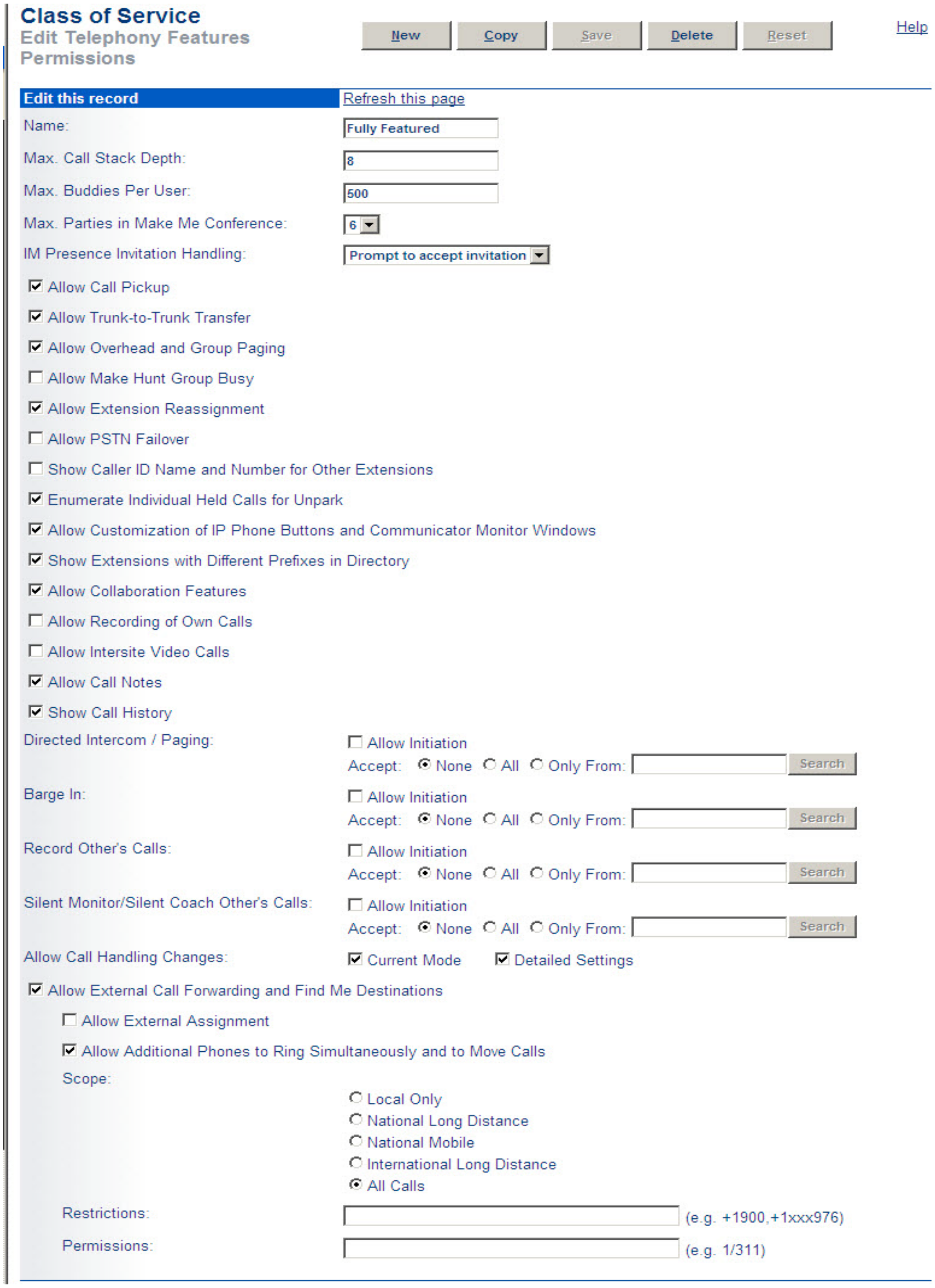


Figure 12-1 **Configuring the Edit Telephony COS window for Extension Assignment**

Step 6 Select the **Allow Extension Reassignment** check box.

Step 7 Select the **Allow External Call Forwarding and Find Me Destinations** check box and the appropriate Scope radio button.

Step 8 Click **Save** to store your changes.

Details:

- If you are intending for a user to have access to the Extension Assignment feature, you should verify that he or she belongs to a User Group that is associated with the Class of Service you just modified above.

12.2.4 Configuring Off-Net External Assignment

The off-net external assignment is intended for mobile users who often work outside the office. These users might travel frequently or work from home, so they could benefit from having all calls directed to an "off-net" device, such as a cell phone or home office PSTN phone. Extension Assignment lets a user have a ShoreTel extension and mailbox without requiring a dedicated switch port and physical telephone in the office.

NOTE For an off-net ShoreTel user's phone to display the ID of a caller who is outside a ShoreTel site, the system administrator must activate the function called Enable Original Caller Information on applicable trunk groups, as introduced in the "Outbound Settings" on page 159 and detailed in the "Forwarding Original Caller ID Outside a ShoreTel Network" on page 174.

To assign an extension to an off-net location, the ShoreTel user does the following steps:

Step 1 Launch **Communicator** on the client machine.

Step 2 Click on the Options menu (the ShoreTel symbol in the upper left hand corner) and select **Extension Assignment > Configure Extension Assignment**.

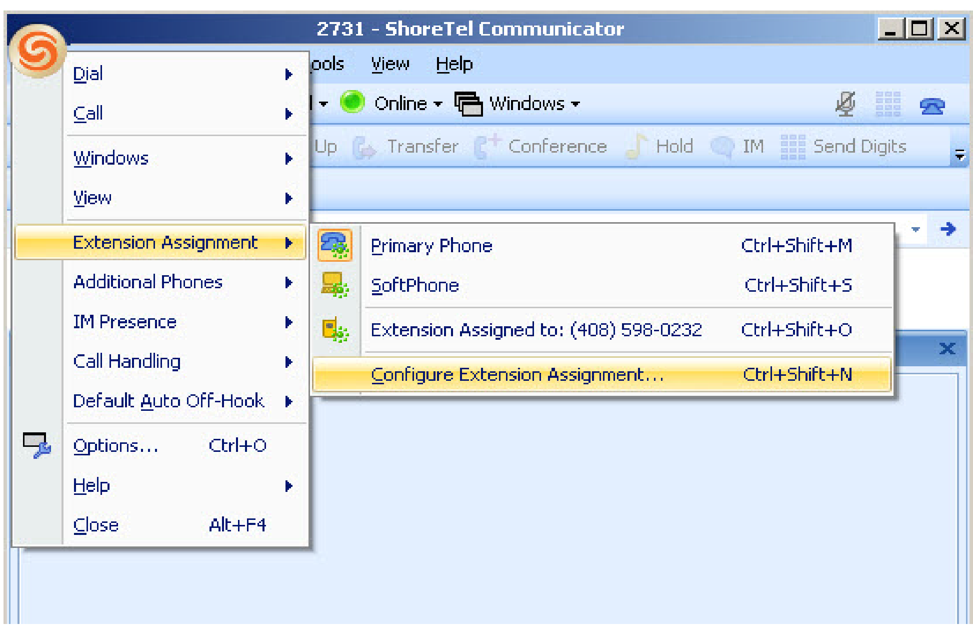


Figure 12-2 Using Communicator to Configure the Client Machine for Extension Assignment

Step 3 A pop-up window similar to the one shown in Figure 12-3 appears.

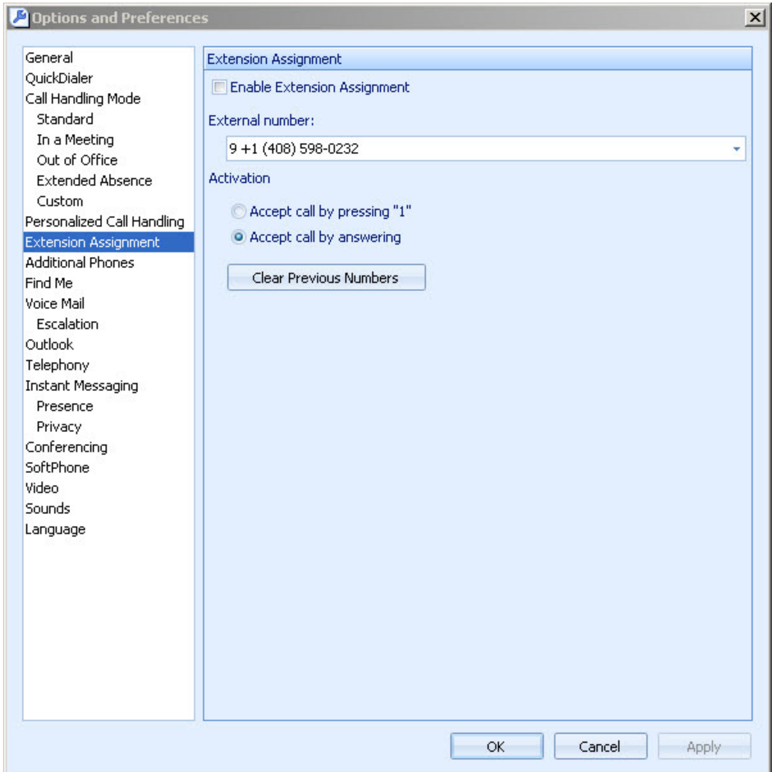


Figure 12-3 Select the Enable External Assignment Check Box

Step 4 Select the **Enable External Assignment** check box

Step 5 Type the external phone number to which calls are to be routed. (This can be a cell phone, home phone, or other PSTN number outside the ShoreTel system.) If the external number has not already been entered, you do it now.

You cannot configure a PSTN number that would require the call to go out a SIP trunk.

Step 6 Select one of the following Activation radio buttons:

- Select **Accept call by pressing "1"** to answer calls on the device by pressing 1.
- Select **Answer call by answering** to answer calls on the device by releasing the switch (e.g. lifting the handset off the cradle).

Step 7 Click the **OK** button to store changes.

Step 8 Click the Options menu and select **Call Handling** > **Configure Call Handling**. A Call Handling window appears as shown in Figure 12-4.

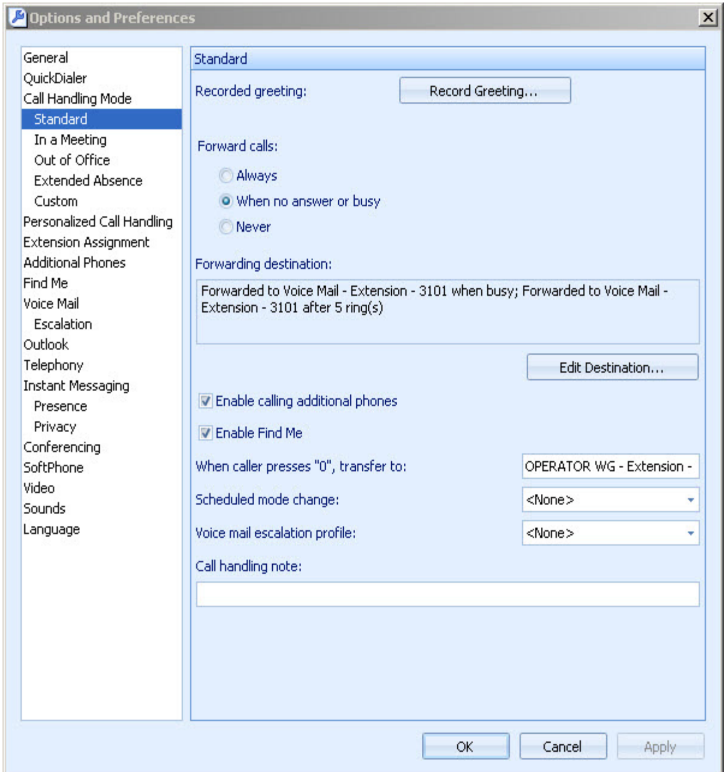


Figure 12-4 Call Handling Window

Step 9 In the Forward Calls area of the window, select the **When No Answer** or **Busy** radio button.

Step 10 Under the “Forwarding destination” area, click the **Edit Destination** button. The Call Handling Destination dialog box appears as shown in Figure 12-5.

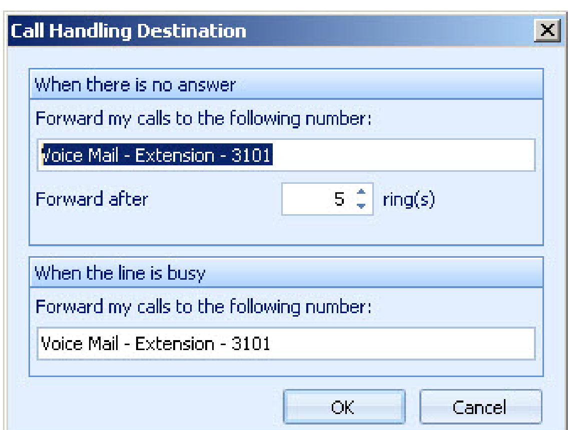


Figure 12-5 Edit Call Handling Destination Window

Step 11 In the “Forward after” field, set the number of times the phone is to ring before the call is sent to voicemail.

NOTE The default of three rings might be insufficient for answering a call before it goes to voicemail. Selecting the Edit Destination button shown in the middle-right area of Figure 12-4 opens a window for modifying the destination and number of rings parameters (Figure 12-5).

Step 12 Click **OK**.

Details:

- When finished with the configuration, the user can verify that Extension Assignment is enabled on the client machine by checking the Forwarding Destination box in the call handling configuration window.

12.2.5 Configuring On-Net Extension Assignment

On-net extension assignment is intended for users who may travel frequently, accessing the ShoreTel system from multiple sites "on the network." These users can benefit from not being tied down to a single physical telephone, and Extension Assignment allows them the freedom to pick up a ShoreTel telephone and assign their extension to that telephone via the voicemail menu.

To assign or unassign an extension to any on-net telephone using the voicemail Telephone User Interface (below) or using Communicator (see the "Inbound Call Management" on page 418), follow the appropriate procedure below:

12.2.5.1 Using the Telephone User Interface

To assign an extension to a telephone:

Step 1 Log in to voice mail.

Step 2 Press *7* to select Change Mailbox Options.

Step 3 Press *3* to select Re-assign Extension.

Step 4 Press *1* to select Assign.

Step 5 Wait for a dial tone, then hang up.

This option is available only from telephone ports and not trunk ports.

To unassign an extension from a telephone:

Step 1 Log in to voice mail.

Step 2 Press *7* to select Change Mailbox Options.

Step 3 Press *3* to select Re-assign Extension.

Step 4 Press *2* to select **Unassign**.

Step 5 Wait for a dial tone, then hang up.

If no other user is assigned to the home telephone port, the extension automatically reverts back to the home telephone. If another user is assigned to the home telephone port, the extension will be assigned to the SoftSwitch until the home telephone port becomes available. A user can "kick out" the other user from the home telephone port by assigning the extension from their home telephone using the procedure above.

12.2.5.2 Using ShoreTel Communicator

The ShoreTel Communicator login is integrated with Microsoft Windows. If you want to run the ShoreTel Communicator from another workstation, you must exit Windows, and log in by using your credentials. (You must have a valid profile on the workstation and ShoreTel Communicator.)

To return to a home telephone port:

Step 1 Verify the user status displayed in the lower-right corner of Communicator.

Step 2 Click the **Go Home** icon on the status bar, as shown below, or select **Go Home** from the **File** menu.

Figure 12-6 **Extension Assignment Status and Go Home Function – Communicator Status Bar**

12.3 Inbound Call Management

12.3.1 Find Me

Find Me is a call handling method that allows a user to route inbound calls to a specified extension or phone number as an alternative to sending callers to voicemail.

Find Me call handling provides inbound callers with a method of connecting to their intended call recipient while listening to the recipient's voicemail greeting. Callers are routed to an extension or phone number callers by pressing "1" while listening to the recipient's voicemail greeting.

System users configured for Find Me can specify two numbers for receiving call through Find Me. The standard voice mail greeting does not prompt the caller on the availability of Find Me call handling.

Find Me call handling is enabled through call handling mode settings. Find Me destinations are independent of the call handling modes that activate Find Me.

After the caller presses "1", the system informs the caller that Find Me destinations are being called. Calls not answered at either Find Me destination are sent to voice mail.

When a call is forwarded to a Find Me destination, the phone at the Find Me destination displays recipient's voice mail caller ID to the call originator. When answering a call, the recipient hears a prompt announcing the call and, if available, the caller's ID information. The recipient can then accept the call or route the caller to voicemail.

Announced Find Me provides for the recording of the caller's name for calls routed to Find Me destinations. This feature provides the capacity for all inbound callers to be identified when their call is routed to a Find Me destination.

When Announced Find Me is enabled, callers from external numbers or from internal extensions without a recorded name are prompted to record their name before the call is routed to the recipient.

12.3.1.1 Find Me and External Assignment Page

When you click the Find Me and External Assignment link on the Personal Options page, the Find Me and External Assignment page appears, as shown in Figure 12-7.

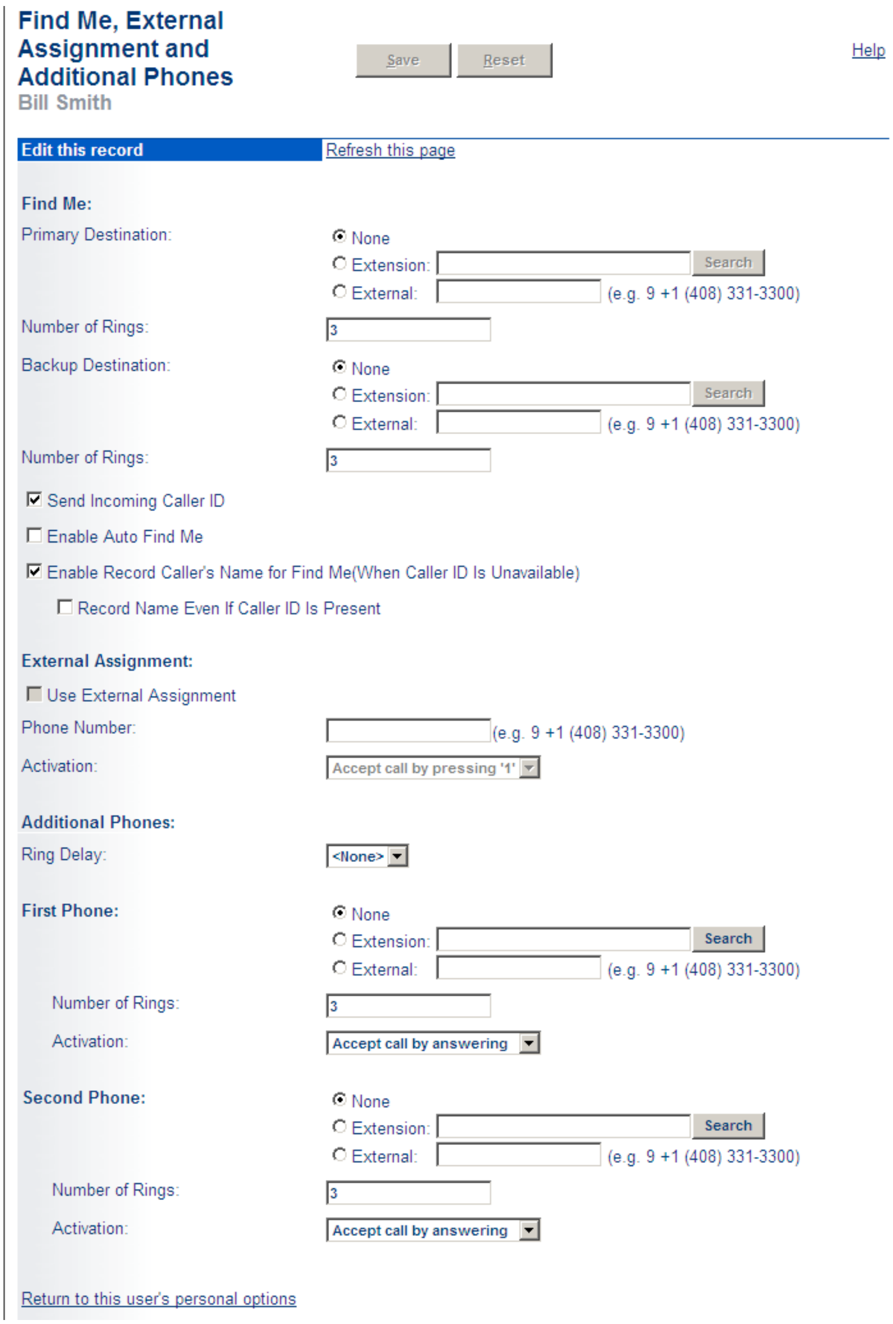


Figure 12-7 Find Me and External Assignment Page

Find Me allow users to configure two phone numbers as forwarding destinations, so that if they miss an incoming call, the call will be sent to voice mail, and from there, it will be redirected to one of the pre-configured forwarding destinations. External Assignment is also known as Extension Assignment. Refer to "Using Extension Assignment" on page 410 for more information.

When a caller dials a user on the ShoreTel system, if the call is not answered, it is sent to voice mail. At this point, the caller is offered the option to “press 1 to activate Find Me call handling,” and the system subsequently attempts to route their call to one of the Find Me destinations that the user configured.

No system prompt informs callers that they need to press 1. The ShoreTel user must inform callers of this “press 1 to activate Find Me” option in the outgoing voice mail greeting.

After pressing “1”, the caller hears a message that the Find Me destinations are being called. If the call is not accepted at either of the Find Me destinations, the call is sent to voice mail.

Alternatively, users also have the option to automate the Find Me behavior, thus bypassing the requirement for callers to “press 1 to activate Find Me.” When Auto Find Me is enabled, calls will be immediately sent to the Find Me destination number(s) without any action on the part of the caller.

Details:

- Users are authorized to use Find Me through a Telephony Class of Service that enables Find Me.
- Find Me destinations can be extensions or external numbers.
- Find Me call handling can be enabled/disabled for each of the five call handling modes. The same Find Me destinations apply to all call handling modes with Find Me enabled.

To configure Find Me destinations:

Step 1 For the primary Find Me destination, select Extension or External. You can use the Search button to find system extensions. Enter external numbers in the External text box.

Step 2 Enter a value in the Number of Rings field. This is the number of times the phone must ring without being answered before the call is forwarded to the primary Find Me destination.

Step 3 For the backup Find Me destination, select Extension or External. You can use the Search button to find system extensions. Enter external numbers in the External text box.

Step 4 Once again, enter a value in the Number of Rings field. This is the number of times the phone must ring without being answered at the primary Find Me destination before being forwarded to the backup Find Me destination.

Step 5 Select the **Send Incoming Caller ID** check box if you want the caller ID forwarded to the Find Me destination.

Step 6 Select the **Enable Auto Find Me** check box to automate the process of routing calls to the Find Me destinations. With this enabled, users will no longer have to press 1 to activate Find Me. You can deselect the check box if you wish to continue to require callers to “press 1 to activate the Find Me feature.”

If you enable Auto Find Me, make sure that your outgoing voice mail message is no longer telling callers to "press 1 to activate the Find Me feature." The automation aspect of this enhancement means that callers will not have to do anything and calls will be forwarded automatically.

Step 7 Click Save.

12.3.1.2 Configuring Announced Find Me

Announced Find Me is available for users assigned to a class of service for which Find Me is authorized. ShoreTel provides three methods of enabling Announced Find Me for a user: from Director, from Communicator for Windows, and from Communicator for Web.

Users can receive Announced Call Me introductions for all callers or restrict the introductions to calls for which Caller ID is not available.

Administrators enable Announced Find Me for system users from the Find Me and External Assignment page.

To enable Announced Find Me for a user, perform the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Users > Individual Users**. The I**ndividual Users list** page appears.

Step 3 Select the user whose profile you want to edit. The **Edit User** page appears.

Step 4 Access the user's Find Me and External Assignment page, as shown in Figure 12-7 on page 419, by selecting Personal Options in the selection bar at the top of the Edit User page, then clicking Find Me at the bottom of the Personal Options page.

Step 5 Select the **Enable Record Caller's Name for Find Me** (When Caller ID is unavailable) checkbox.

Step 6 To enable Announced Find Me when Caller ID is available, select the **Record Name Even If Caller ID Is Present** checkbox.

12.3.2 Automated Call Handling

A Call Handling Mode defines call management conditions and tasks for your inbound calls. ShoreTel defines the following five call handling modes for each extension to customize the manner that a user's calls are handled over a variety of situations:

- Standard.
- In a Meeting.
- Out of Office.
- Extended Absence.
- Custom.

One call handling mode is always active. ShoreTel automatically selects the active call handling mode on the basis of system schedules maintained by the system administrator. Users can also manually select their active call handling mode.

The following sections describe call management tasks controlled by Call Handling Modes, along with the automatic and manual methods of selecting the active call handling mode.

12.3.2.1 Setting the Active Call Handling Mode

The active Call Handling Mode specifies the manner in which a user's inbound calls are handled. ShoreTel automatically activates call handling modes, as specified by system schedules maintained by the administrator. Users can also manually select the active call handling mode.

A ShoreTel schedule is a software component that specifies a set of time periods. Communicator uses schedules to program call handling mode changes for system users.

Each schedule consists of a name and a list of day-time intervals. Communicator pages refer to the schedule by its name and specify an active call handling mode during the intervals.

ShoreTel defines three schedule types:

- **On-hour schedules:** On-hour schedules specify day and time intervals over a weekly period without referencing a specific date. Figure 12-8 displays three examples of On-hour schedules.

Day Shift		Afternoon Shift		Weekend Shift	
Monday	8:00 am - 5:00 pm	Monday	1:00 pm - 10:00 pm	Saturday	8:00 am - 12:00 pm
Tuesday	8:00 am - 5:00 pm	Tuesday	1:00 pm - 10:00 pm	Sunday	8:00 am - 12:00 pm
Wednesday	8:00 am - 5:00 pm	Wednesday	1:00 pm - 10:00 pm		
Thursday	8:00 am - 5:00 pm	Thursday	1:00 pm - 10:00 pm		
Friday	8:00 am - 5:00 pm	Friday	1:00 pm - 10:00 pm		

Figure 12-8 On-Hour Schedule Examples

- **Holiday schedules:** Holiday schedules specify dates. Periods that identify a year are valid once; periods that do not specify a year are valid each year on the listed dates. Holiday schedules take precedence over On-hour schedules during periods covered by each schedule. Figure 12-9 display examples of Holiday Schedules.

Holiday 1	Holiday 2
1/1	1/1
5/26/2008	2/18/2008
7/4	3/17
11/27/2008	7/4
11/28/2008	9/1/2008
12/25	11/27/2008
	11/28/2008
	12/25

Figure 12-9 Holiday Schedule Examples

- **Custom schedules:** Custom schedules specify date and time intervals. Periods that identify a year are valid once; periods that do not identify a year are valid each year on the listed dates. Custom schedules take precedence over On-hour and Holiday schedules during periods listed by multiple schedules. Figure 12-10 display examples of Holiday Schedules.

Custom 1		Custom 2	
6/14/2008	9:00 am - 4:00 pm	2/5/2008	12:00 am - 12:00 pm
7/24	6:00 am - 9:00 pm	2/6/2008	12:00 am - 12:00 pm
		2/7/2008	12:00 am - 12:00 pm

Figure 12-10 Custom Schedule Examples

Schedules are configured by the system administrator. To determine the periods specified by an individual schedule, consult your system administrator.

Default active call handling modes are managed through the Schedule parameter of the Call Handling Mode configuration pages:

- The **Standard** and **Out of Office** modes determine the active call handling mode during normal daily activities.
  - — The available options in **Standard**'s schedule parameter are the *On-hour* schedules configured by the system administrator. **Standard** is the default Call Handling Mode during the times listed by the selected schedule.
  - — The **Out of Office** schedule is the inverse of the selected schedule in the **Standard** Call Handling Mode configuration page. Out of Office is the default Call Handling Mode during all periods not covered by the On-hour schedule in the Standard Call Handling Mode page. Schedules are not selected in the Out of Office Call Handling Mode page.

    Example: Assume that Day Shift in Figure 12-8 is the selected schedule in the **Standard** Call Handling Mode page. **Standard** is the default call handling mode between 8 am to 5 pm on Mondays through Friday. Out of Office is the default call handling mode all other times.
- The **Extended Absence** mode determines the active call handling mode during the selected holiday schedule. The Schedule drop down menu on the Extended Absence call handling mode lists the Holiday schedules configured by the administrator.
- The **Custom** mode determines the active call handling mode during the selected custom schedule. The Schedule drop down menu on the Custom call handling mode lists the Custom schedules configured by the administrator.
- **In a Meeting** mode is not associated to a schedule and cannot become active through an automatic mode selection.

12.3.2.2 Call Handling Mode Defaults

Call Handling Mode Defaults are the set of call handling parameters assigned each time you add a new user. ShoreTel strongly recommends that you review and change these defaults before you add the bulk of your users.

Once a user is saved on the system, there is no relationship between the user's call handling modes and the default call handling modes. Changes to the default call handling modes do not affect the call handling modes of current users.

If you need to change the Personal Assistant of some or all users, you can use the Batch Update Utility.

There are five default call handling modes, used for initializing each user's call handling modes. These modes provide a quick and easy way for users to change the way their inbound calls are handled.

The five default call handling modes are:

- Standard.
- In a Meeting.
- Out of Office.
- Extended Absence.
- Custom.

Each of the call handling modes has the same configuration parameters.

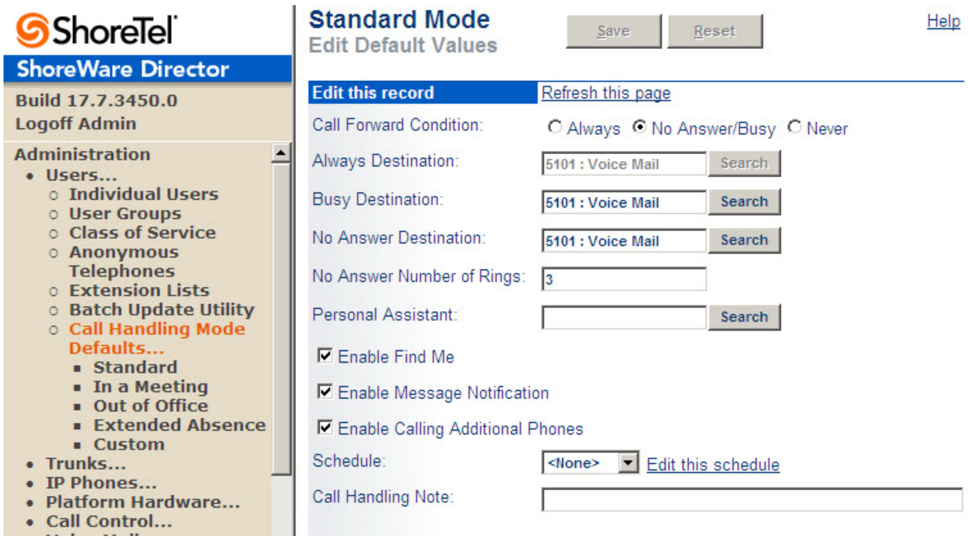


Figure 12-11 Call Handling Mode Default Values Edit Page

Call handling modes specify how, when, and where calls are forwarded, and whether the user requires message notification when voice mail is received. The links in the Edit Call Handling Modes section under the Personal Options tab on the Edit User page bring up variants of the call handling mode pages. You can edit these copies for each user's personal options. Users can also modify these options from their desktop client applications.

Users can also change their call handling settings though a web interface on the ShoreTel server. For more information and the web interface URL, see the ***ShoreTel 12: Planning and Installation Guide***.

The Call Handling Mode Default parameters are as follows:

- **Call Forward Condition:** These buttons let you specify when calls are forwarded. The conditions are Always, No Answer/Busy, and Never.
  - — The **Always** condition forwards calls to the number specified in the Always Destination parameter immediately when a call is received.
  - — The **No Answer/Busy** condition forwards calls to the No Answer Destination after the specified number of rings, or to the Busy Destination immediately if the user's call stack is full.
  - — The **Never** condition disables call forwarding.

  The recommended default is No Answer/Busy.
- **Always Destination:** When the Always call forward condition is selected, calls are forwarded immediately to this extension.
- **Busy Destination:** When the No Answer/Busy call forward condition is selected, calls are forwarded to this extension immediately if the user's call stack is full.

The recommended default is Voice Mail.

- **No Answer Destination:** When the No Answer/Busy call forward condition is selected, calls are forwarded to this extension after the specified number of rings.

  The recommended default is Voice Mail.

- **No Answer Number of Rings:** When the No Answer/Busy call forward condition is selected, this parameter specifies how many times the phone rings before the call is forwarded to the No Answer Destination.

  The recommended default is three rings.

- **Personal Assistant:** Each user can specify a Personal Assistant, which is the destination to which a calling party is transferred upon dialing "0" in the user's mailbox. For example, executives often want callers transferred to their own executive assistant rather than to the operator when a caller dials "0" in their mailbox.

  If no personal assistant is defined and a caller dials "0," the call is transferred to the site operator. If no site operator is defined, the call is transferred to the auto-attendant.

  Users can also reach the Personal Assistant from the voice mail menu. By pressing 0 from the main voice mail menu, users can access the assistant. Alternatively, users can press 00 while listening to a voice mail message to reach the assistant. This can be helpful if a user is checking voice mails and wants to quickly reach the assistant to communicate something heard in a voice message.

  The recommended default is an operator.

- **Enable Find Me:** This check box enables the ***Find Me*** feature by default for new users. When enabled, users can configure up to two numbers where they would like to receive calls that are forwarded from their voice mail. For more information, see the "Find Me and External Assignment Page" on page 419.

- **Enable Message Notification:** This check box enables message notification for this call handling mode. The manner in which the user is notified is determined by the user's message notification settings.

  The recommended default is off.

- **Schedule:** This drop-down menu lets you select the schedule that will be associated with this call handling mode. For example, you may want to associate the Standard CHM with a schedule that is active from the hours of 9am to 5pm. Alternatively, you might wish to schedule the Out of Office CHM with a "graveyard" schedule that becomes active from the hours of 10pm to 6am.

  For more information, see the Chapter 15: Configuring Schedules starting on page 475.

- **Call Handling Note:** This text entry option lets you type a note that is visible only to users whose access license is Operator. We recommend leaving the field empty.

12.3.3 Personalized Call Handling

Personalized Call Handling defines a method of handling inbound calls on the basis of multiple end user, caller, system, and environment criteria.

Personalized Call Handling is available to all System Users that are authorized for the following Communicator types:

- Professional.
- Operator.
- Workgroup Agent.
- Workgroup Supervisor.

Administrator authorization is not required for users assigned to these Communicator types.

12.3.3.1 Description

A Call Routing Plan manages a user's inbound voice calls. The plan consists of call handling rules, each of which specifies a method of handling a call when a condition set is valid. The plan enables and prioritizes selected rules. The user's inbound calls are evaluated against the call handling rules. The highest priority rule with conditions that are satisfied define the handling method for the call.

This section describes the structure and call handling rules, followed by a discussion of creating a call routing plan from these rules.

Call Handling Rules

Call handling rules define evaluation conditions and handling action. When the rule is active and the condition is satisfied, the action determines how the call is handled.

Three components comprise a call handling rule: name, condition, and action. The following sections describe call handling rule components.

- **Name:** The name is the label by which Communicator and Director refers to a call handling rule. Users specify the name when they create the rule.
- **Condition:** The condition is the filtering criteria that determines if a corresponding call handling action is performed. When a condition statement consists of multiple criteria, each criterion must be satisfied or the action is not performed.

  Call Handling Rules defines the following six types of criteria:

  — **Phone number match:** The phone number match is satisfied when the caller ID of the inbound call is a subset of the specified match type. Users can select one of the following match types:

    - **Specific number –** user specifies the number that must match the caller ID; numbers can be internal extension or phone number external to the system.
    - **Off system extension –** user specifies the off system extension that must match the caller ID.
    - **Any internal extension starting with –** user specifies the digits that must match the first digits of numbers originating from internal callers.
    - **Any external extension starting with –** user specifies the digits that must match the first digits of numbers originating from phone numbers external to the system.
    - **Private –** matches all calls that are identified by caller ID as private.
    - **Out of area / unknown –** matches all calls that caller ID identifies as out of area or unknown.
    - **Every internal number –** matches all calls originating from a device within the ShoreTel network.

    - **Every external number –** matches all calls originating from a device not located within the ShoreTel network.

    Users can specify a maximum of ten phone number match entries.

  - **I am on the phone:** This criterion is satisfied when the user's phone is ***busy***.
  - **Call Handling Mode:** This criterion is satisfied when the user's active Call Handling Mode matches the specified mode.
  - **Time of day:** This criterion is satisfied when the call is received during the specified time ranges.
  - **Day of week:** This criterion is satisfied when the call is received on one of the specified days.
  - **DNIS Match:** This criterion is satisfied when the DNIS of the inbound call matches the specified number.

    Time of day and **Day of week** entries are based on the timezone setting of the site to which the user is assigned.

- **Action:** The action specifies the resolution for calls that match the specified condition. Call Handling Rules define the following five action types:
  - **Forward Call to Specific Number:** This action routes the call to the specified number. Users can select one of the following number types:
    - Specific Number.
    - Off System Extension.
  - **Forward Call to Voice Mail:** This action routes the call to the recipient's voice mail.
  - **Forward Call to Auto Find Me:** This action routes the call to voicemail, which then forwards the call to the recipient's Find Me number.
  - **Forward Call to Announced Find Me:** This action routes the call to voicemail, which then forwards the call to the recipient's Find Me number. The system attempts to announce the caller's name to the recipient, which may require the caller to record their name before the call is presented to the recipient.
  - **Play Ringtone:** This action programs the ShoreTel IP Phone to play the designated ring tone that announces the presence of the inbound call to the recipient.

Call Routing Plan

The Call Routing Plan is the data structure that determines the routing method of a user's inbound calls. The following sections describe the composition and operation of a call routing plan.

A call routing plan consists of a maximum of ten call handling rules. The plan specifies the rules that are active and lists the rules in the order that by which they are used to evaluate characteristics of the inbound call.

When a user receives an inbound call, the system evaluates the characteristics against the highest priority call handling rule that is enabled. If all of the criteria comprising the condition of that rule matches the call characteristics, the call is routed as specified by the rule's action and the plan execution is complete. If any of the criteria do not match the call characteristics, the system continues the plan execution by evaluating the call against the next highest priority call that is enabled.

This process is repeated for all enabled call handling rules. If the call characteristics do not match the conditions of any enabled call handling rule, the call is routed as specified by the active Call Handling Mode.

12.3.3.2 Implementing Personalized Call Handling

Users configure their Personalized Call Handling routing plans through Communicator. Refer to the Communicator User's Manual.

The Director Personalized Call Handling Rules page, shown in Figure 12-12, displays the Call Handling Rules created by the specified user. The table displays the name, condition, and action and enabled status of each rule. The rules are listed in the order that they were created; the table does not display the usage priority of the rules.

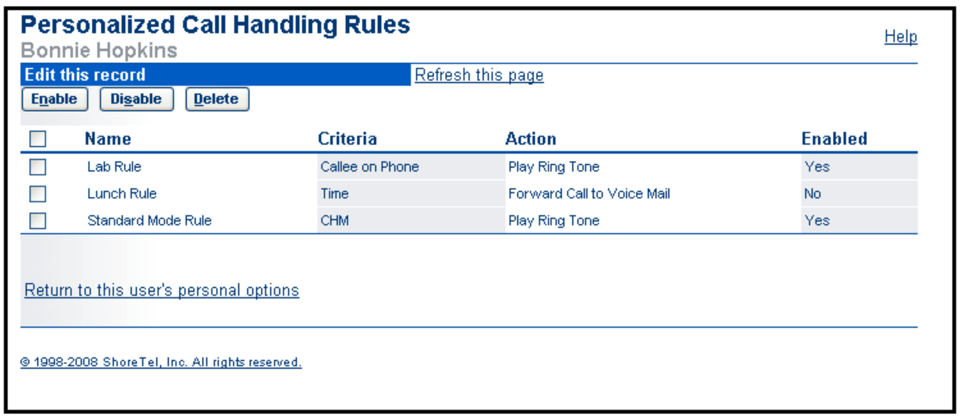


Figure 12-12 Personalized Call Handling Rules Page (Director)

To access the Personalized Call Handling page:

Step 1 Open the Individual User list by selecting **Administration -> Users -> Individual Users** from the Director Main Menu

Step 2 Open the Personal Options page for the desired user by clicking the name of the desired user in the Individual User List, then selecting Personal Options on the menu bar toward the top of the Edit User page.

Step 3 Click Personalized Call Handling Rules, located in the middle of the Personal Options page, as shown in Figure 12-13.

Figure 12-13 Edit User Page – Selecting Personalized Call Handling Rules

12.4 Call Intervention Methods

12.4.1 Whisper Page

The Whisper Page feature allows a user to break into an active call in order to speak with an internal user. This occurs without the remote caller hearing the interruption and without the operator hearing the remote caller.

A real-world example illustrates the function: You are on a call with a client when another client arrives in the lobby for an appointment with you. The administrative assistant knows that you are on a call and uses the Whisper Page feature to interrupt the call to announce that someone is waiting for you in the lobby. You hear the voice of the administrative assistant and the client at the same time, but neither of them can hear the other.

Implementation details:

- The Whisper Page feature can be invoked from:
  - — Communicator
  - — Any phone (analog or IP) by pressing the code *19
  - — One of the ShoreTel IP Phone soft keys
- While on a Whisper Page call, the internal user can mute the audio channel to the original caller. The user can respond to the operator without the original caller hearing. This can be accomplished from:
  - — One of the ShoreTel IP Phone soft keys rather than the standard mute button
  - — Communicator, if you do not have an IP phone

- Both the operator and the internal user hear a tone when the Whisper Page call is connected. (The tone is the same as the tone for the Intercom feature.)
- To receive a Whisper Page call, the internal user must be on the handset of a multiline ShoreTel IP Phone. If a Whisper Page call is sent to any other phone (SoftPhone) the call will be treated as an intercom call.
- If a Whisper Page call is sent to a phone that is not on an active call, the feature behaves the same as an intercom call.
- The Whisper Page feature does not work if the internal party is on a three-way conference call.
- No call control operations can be performed on a Whisper Page call except to hang up the call. For example, the Whisper Page call cannot be put on hold, transferred, parked, and so on.

Configuring Whisper Page

To configure the Whisper Page feature:

Step 1 Launch ShoreTel Director and enter the user ID and password.

Step 2 Click **Administration > Users > Class of Service**. The Class of Service page appears.

Step 3 Select a Telephony Features Permissions profile. The Edit Telephony Features page appears, as shown in Figure 12-14.

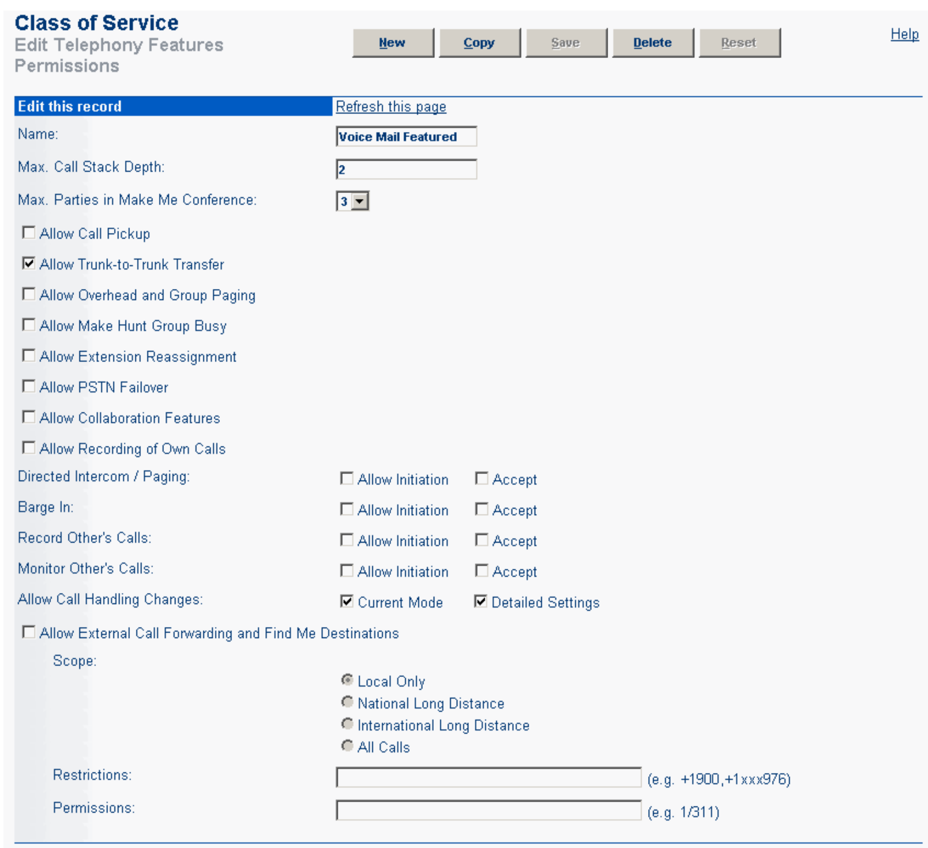


Figure 12-14 **Edit Telephony Features Page**

There is no separate check box specifically for the Whisper Page feature. The functionality has been coupled with the intercom functionality.

Step 4 In the Whisper Paging field, check the **Allow Initiation** check box to allow this user to place a Whisper Page call. (This user would most likely be an operator or administrator.) The Accept check box enable lets a user be on the receiving end of a Whisper call.

Step 5 Click **Save** to save changes.

After the Whisper Page feature in a Class of Service profile is enabled, be sure that this Class of Service profile is applied to the appropriate user group to which the target user belongs. Put another way, the user intended to use the Whisper Page feature must belong to a user group that has the COS - Telephony value set to the name of the COS profile.

12.4.2 Monitoring Extensions from an IP Phone

Extension monitoring is one of the available functions you can assign to the custom buttons on a ShoreTel IP Phone. However, due to its complexity, this feature is described separately from the other functions.

The extension monitoring function lets a user monitor the extension of another user and answer the other person's calls if necessary. For example, two secretaries are working on different floors of the same building and are responsible for answering calls from the main phone line. If one secretary already is on a call when another call arrives on the main line (with extension monitoring enabled), the other secreatry can see that the first assistant is busy and, therefore, knows to answer the incoming call.

To configure extension monitoring:

Step 1 Navigate to **Administration > Users > Individual Users** in ShoreTel Director.

Step 2 Click on a name to modify that user’s phone. The user’s general settings appear.

Step 3 Click the **Personal Options** tab.

Step 4 Click the link for **Program IP Phone Buttons**. The Program IP Phone Buttons editor opens, as Figure 12-15 illustrates.

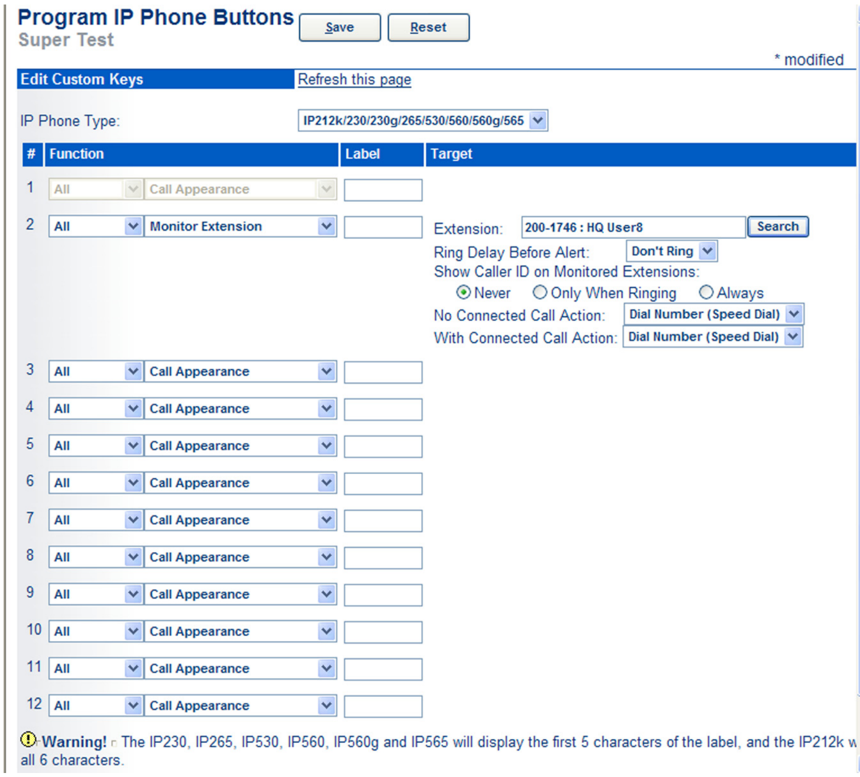


Figure 12-15 Monitor Extension from IP Phone

Step 5 For the intended button (except number 1, which is reserved), click on the **Function** drop-down menu for one of the programmable buttons and select **Monitor Extension** to display the custom key editor.

Step 6 Enter a label for the button in the Label field.

Step 7 Enter the extension of the monitored party in the Extension field, or click the Search button to locate the desired extension.

Step 8 Click on the **Ring Delay Before Alert** drop-down menu and select the number of rings that should elapse on the monitored party's phone before the monitoring party will receive an alert. (This delay gives the monitored party a chance to answer and prevents the monitoring party's phone from ringing incessantly.)

Step 9 Select the appropriate radio button in the Caller ID on Monitored Extensions. For an multiline phone, choices are:

- **Do Not Show –** The Caller ID information does not appear, but an indicator will show that the monitored phone is busy. This option offers the monitored party the most privacy and should be selected if you do not want the monitoring party to know who the monitored party is talking to.
- **Show Only When Ringing –** The Caller ID information appears while the phone is ringing, but disappears once the call has been answered.

- **Show Always –** The Caller ID information appears while the monitored phone is ringing, and continues to appear even after the call has been answered.

Step 10 The custom buttons can be configured to perform different actions based on whether or not the person being monitored is on a call. To associate a secondary function with the custom button that will apply when the phone is inactive, click the drop-down menu and select another action from the ***No*** Connected Call Action menu. The action you select here will apply when the custom button is pressed AND while the monitoring party's phone is inactive.

Step 11 To associate a third function with the custom button, click the drop-down menu and select another action from the **With Connected Call Action** menu. This action will apply when the custom button is pressed while the monitoring party does not have a call that can be picked up or unparked, and the user's own extension has a connected call.

The idea behind these last two steps is that the custom button can be configured to do whatever makes sense given the situation. The button can be configured to pick up incoming calls, or park/unpark calls when the person being monitored is on a call. And if the person being monitored is not on a call, the custom button then becomes a one-touch button allowing the operator to transfer a call to the monitored extension. And if the person being monitored is not on a call and if there are no calls to transfer, then the custom button becomes a speed-dial button, allowing the operator to dial that person's extension at the touch of a button.

Step 12 Click **Save** to store your changes.

Details:

- The custom button will illuminate red (on the monitoring-person's phone) when the person being monitored is on a call. If that call is put on hold and a second call is accepted on the monitored extension, the LED will turn green and will flash twice. Similarly, the LED will flash three times if a third call is accepted. For more information about LED flash patterns, see Table 12-1.
- The custom button (to which extension monitoring has been assigned) can serve dual purposes based on whether the monitoring party is in a call or not. The button can be set to speed dial, intercom, or transfer calls to the monitored extension.
- When the "Show Caller ID Name and Number on Monitored Extensions" Class of Service (Telephony) setting is not enabled, Communicator Contact Viewer (and Agent Viewer) show the number of calls on a user's stack but do not show who the user is talking to. "Properties" is also disabled.

Table 12-1 Programmable Buttons LED Flash Patterns

State	Pattern
CALL APPEARANCE STATES	
Idle	Off
Idle and DND	Orange, Steady On
Idle and Message Waiting	Off
Idle, Message Waiting and DND	Orange, Steady On

Table 12-1 Programmable Buttons LED Flash Patterns (Continued)

State	Pattern
Off Hook	Green, Steady On
Active Call	Green, Steady On
Active Conference Call	Green, Steady On
Remote Hold	Green, Steady On
Offering Call	Green, 1000/1000 ms
Held or Parked Call (3)	Orange, 250/250 ms
Whisper Page Call	Red, Steady On
Active Call Whisper Muted	Red, Steady On
EXTENSION MONITOR STATES	
Idle	Off
Idle and DND	Orange, Steady On
Idle and Message Waiting	Off
Idle, Message Waiting and DND	Orange, Steady On
Offering Call	Green, 1000/1000 ms
Active Call Picked Up	Green, Steady On
Held or Parked Call [3]	Orange, 250/250 ms
Monitored Ext. on Active Call	Red, Steady On
Monitored Ext. on Conference Call	Red, Steady On
Monitored Ext on Active Call + Offering Call	Green, 200/100/700/1000 ms
Picked up Monitored Ext. Call + Monitor Ext on Active Call	Green, 800/Orange 200 ms
Picked up Monitored Ext. Call and Held + Monitor Ext on Active Call	Orange, 200/100/200/500 ms
Picked up Monitored Ext. Call + Monitor Ext held Active Call	Orange, 200 ms Green, 800 ms Orange, 200 ms Green, 100 ms
BRIDGED CALL APPEARANCE STATES	
Idle	Off
Offering Call	Green, 1000/1000 ms
Active Call Picked Up	Green, Steady On
Line In-Use	Red, Steady On
Held or Parked Call [3]	Orange, 250/250 ms
FEATURE KEY WITH EXTENSION TARGET STATES	
Idle or Offering Call	Off

Table 12-1 Programmable Buttons LED Flash Patterns (Continued)

State	Pattern
Connected or Held Call	Red, Steady On
DND	Orange, Steady On
(Dial/Transfer Mailbox Only) MWI	Red, Steady On
(Pickup, Pick/Unpark, Pickup NightBell Only) Offering	Green, 1000/1000 ms
(Unpark, Pick/Unpark Only) Held/Parked	Orange, 250/250 ms
TOGGLE FUNCTIONS (RECORD, WHISPER MUTE)	
Function Off	Off
Function Available	Orange, Steady On
Record Active	Orange, 500/500 ms
Whisper Mute Active	Orange, 500/500 ms

12.4.3 Call Handling Mode Delegation

The Call Handling Mode Delegation window lets the system administrator specify a list of users who can change the Call Handling Mode of another user. A delegated (authorized) user with an Operator Access License can modify another user's Call Handling Mode through ShoreTel Communicator or Communicator for Web Client. These authorized (delegated) users are specified through ShoreTel Director by the system administrator or through Communicator by the user whose active Call Handling Mode is changed.

12.4.3.1 Delegating through ShoreTel Director

To select the users who are authorized to change another user's active call handling mode:

Step 1 Navigate to **Administration > Users > Individual Users**.

Step 2 Select the user for whom you are authorizing other users to change the active call handling mode of the selected user.

Step 3 Click the **Personal Options** tab.

Step 4 In the Call Handling Mode Options section, click **Delegation**. The link is next to the Current Call Handling Mode field. The Call Handling Mode Delegation window opens (Figure 12-6).

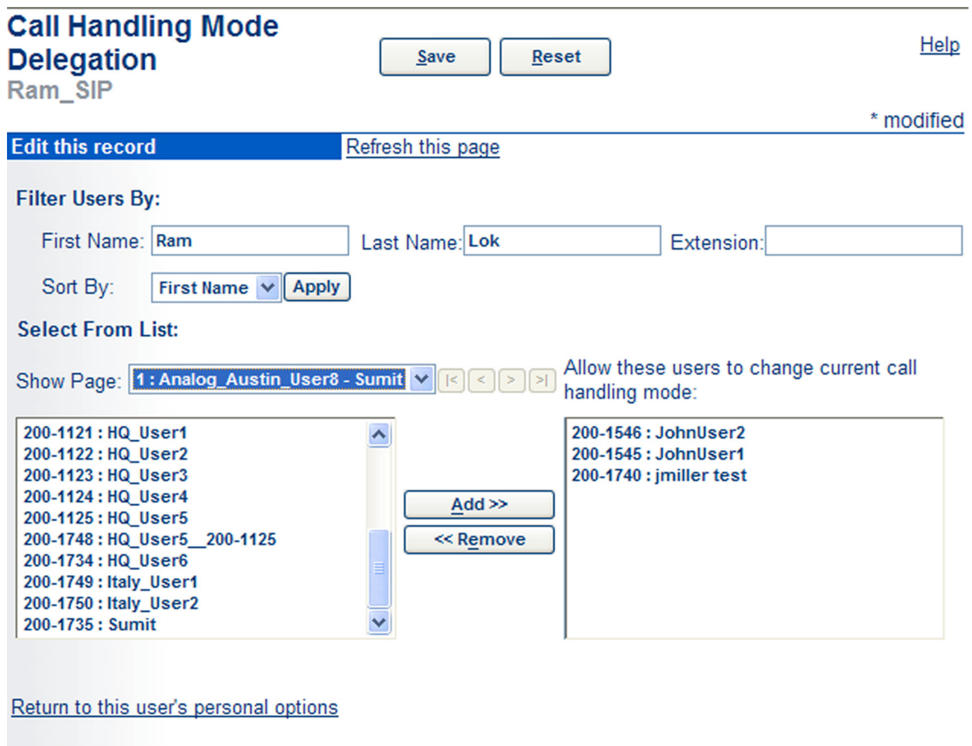


Figure 12-16 Delegating Call Handling Mode to Other Users

Step 5 In the left column, select one or more users to authorize.

Step 6 Click **Add** (or Remove if you are removing names from the list of delegates).

Step 7 Click **Save** at the top of the window when delegation is complete.

12.4.3.2 Selecting Delegates through ShoreTel Communicator

Users can manage the list of individuals who can change their active Call Handling Mode through Communicator. The *ShoreTel Communicator for Widows Guide* contains more information.

12.4.4 Intercom, Whisper Paging, Barge In, Record, and Monitor

12.4.4.1 Feature Descriptions

Monitoring allows one party to eavesdrop on a call. It is a limited conference call where the monitoring party hears the other parties, but the monitored parties do not hear the monitoring party. Monitoring is undetectable by the parties being monitored, except by a warning tone. Monitoring is typically used in workgroups to evaluate agent performance.

A recording warning tone may be played to the customer during call recording and monitoring. The warning tone is enabled for the entire system using a Call Control option. For a specific call, it may be disabled by using an Auto-Attendant menu option. No tone is played during a Barge In call.

Barge In allows one party to join an existing call as a fully conferenced participant. When Barge In is initiated, a brief intrusion tone is played to the other participants and (if present) the monitoring warning tone is discontinued.

To simplify discussion of this feature, we will refer to three parties: the supervisor, the agent, and the customer. The supervisor initiates monitoring by selecting an agent. The agent is on a call with another party, the customer. The customer may be an external caller, but supervisors and agents must be on extensions.

In a monitored call, a supervisor hook flash is ignored. However, a hook flash by the other parties works the same as in a two-party call. In particular, an agent flash puts the call on hold and allows a consultative transfer or conference.

Because there is a limit of three parties in a conference call, if the agent or customer makes a consultative transfer or conference, the supervisor is automatically dropped. Similarly, if another party barges into a monitored extension, then the monitoring is dropped.

If a conference call is already in progress, it cannot be monitored. If monitoring is already in progress, no one else can monitor the call.

The supervisor can barge into a call he or she is monitoring. However it is not possible to revert a barge in to just a monitored call. If desired, the supervisor can hang up and restart monitoring.

After a barge in, the agent remains the controlling party of the call. A subsequent agent hook flash disconnects the supervisor, who was the last party added.

12.4.4.2 COS Settings

Each telephony COS permission has check boxes and radio buttons in ShoreTel Director for configuring Intercom, Whisper Paging, Barge In, Call Recording, and Call Monitoring. These options appear near the bottom of the window in Figure 12-17. For more information about setting permissions, refer to "Telephony Features Permissions" on page 315.

Figure 12-17 The COS Edit Telephony Features Permissions Window

- **Allow initiation for Intercom/Paging**—If this check box is selected, users with this COS can place an intercom call or page to other system users. If this box is empty, the user with this COS cannot initiate an intercom call or page.
- **Accept Intercom/Paging**—Radio button choices are:
  — **Accept: None** If the choice for Accept is None, users with this COS cannot receive intercom calls or pages.
  — **Accept: All** If the choice for Accept is All, users within this COS can receive intercom calls or pages from anyone in the COS.
  — **Accept: Only From:** If the choice for Accept is Only From, users with this COS may receive intercom calls or pages from only the person or extension specified in the associated field.
- **Allow initiation for barge in**—If this check box is selected, users within this COS may barge in on the calls of other system users. If cleared, then no barge in can be initiated.
- **Accept barge in**—Radio button choices are:
  — **Accept None:** If selected, users within this COS may not receive barge-in's from anyone.

- **Accept All:** If selected, users within this COS may receive barge-in's from anyone else in this COS.
- **Accept Only From:** If selected, users within this COS may only receive barge-in's from the person or extension specified in the field associated with this radio button.

- **Allow initiation for record others calls—**If this check box is selected, users within this COS may record the calls of other system users. If cleared, then no call recording can be initiated.
- **Accept record others calls—**Radio button choices are:
  - **Accept None:** If selected, users within this COS may not have their calls recorded from anyone.
  - **Accept All:** If selected, users within this COS may have their calls recorded from anyone else in this COS.
  - **Accept Only From:** If selected, users within this COS may only have their calls recorded by the person or extension specified in the field associated with this radio button.
- **Allow initiation for silent monitor—**If this check box is selected, users within this COS may monitor other system users. If cleared, then no monitoring can be initiated.
- **Accept silent monitor—**Radio button choices are:
  - **Accept None:** If selected, users within this COS cannot be monitored by anyone.
  - **Accept All:** If selected, users within this COS can be monitored by anyone else in this COS.
  - **Accept Only From:** If selected, users within this COS can only be monitored by the person or extension specified in the field associated with this radio button.

No special permissions exist for ShoreTel Enterprise Contact Center agents or supervisors. However, to use center recording, monitoring, and barge in, an agent or supervisor must have a CoS with the settings that allow these features.

CHAPTER 13

Configuring Voice Mail

This chapter provides information about configuring the voice mail system. The topics are:

- "System Distribution Lists" on page 441
- "AMIS Voice Mail" on page 443
- "Delivery and Notification" on page 448
- "Voice Mail Reports" on page 456
- "Voice Mail Synchronization with Gmail for Business" on page 459

Also refer to the "Voice Mail Permissions" on page 323 for information about configuring your voice mail system.

13.1 System Distribution Lists

System distribution lists provide a mechanism for sending the same message to multiple users at one time. They are managed from the ShoreTel Director. You add and edit system distribution lists from the Voice Mail System Distribution Lists. To access the Voice Mail System Distribution Lists page, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Voice Mail > System Distribution Lists**. The System Distribution Lists page appears as shown in Figure 13-1.

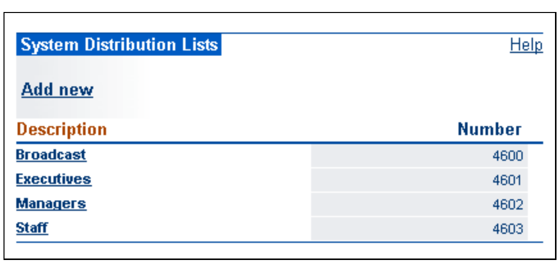


Figure 13-1 Voice Mail System Distribution Lists Page

NOTE You can also add or remove a user from a system distribution list from the Edit User page (see the "Individual Users" on page 329).

The columns in the System Distribution List page are as follows:

- **Description:** This is the name of the system distribution list.
- **Number:** This is the number that is used for sending messages to members in the distribution list. Users can enter this number in either the ShoreTel client or when addressing a message from the telephone user interface.

The ShoreTel system lets a user with the proper class of service send a broadcast message to all mailboxes. Unlike system distribution lists, the broadcast distribution list cannot be edited. If necessary, the administrator can remove individual mailboxes from the broadcast list on the associated Edit Users page, Workgroup edit page, or Route Point edit page.

To add or edit a system distribution list, click "Add new" or click the name of an existing list that appears in the Voice Mail System Distribution Lists table. When you click either of these items, the Edit System Distribution List page appears as shown in Figure 13-2.

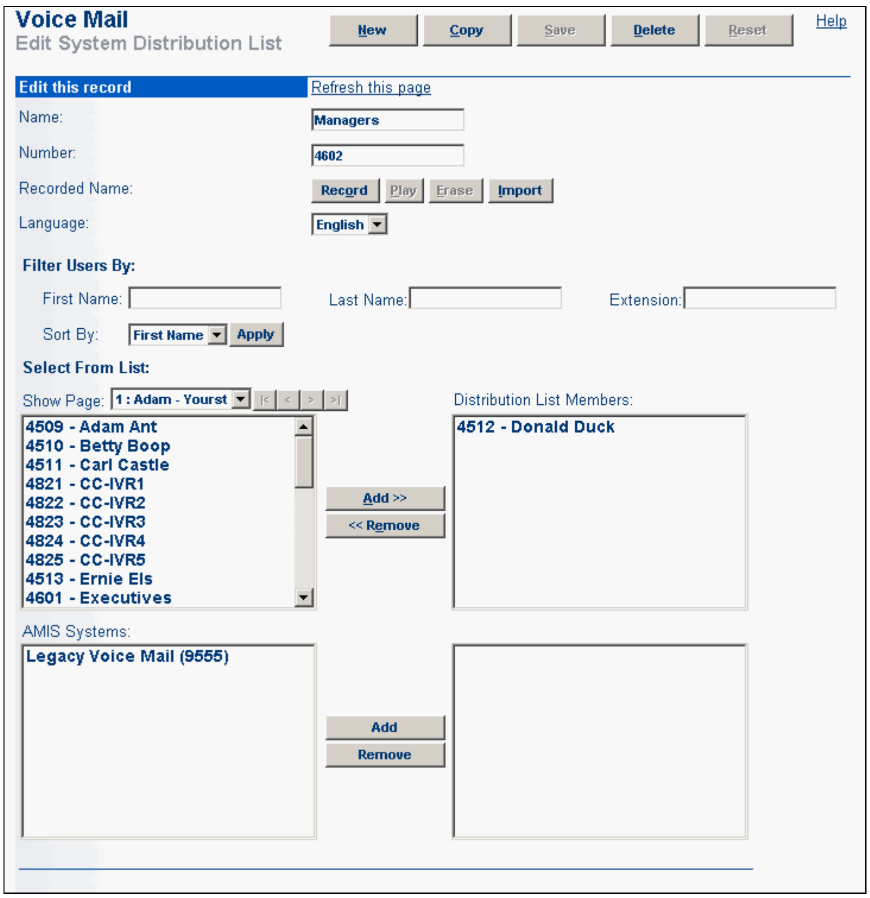


Figure 13-2 System Distribution List Edit Page

The parameters on the System Distribution List edit page are described as follows:

- **Name:** This is the name of the distribution list.
- **Number:** This is the number that is used for sending messages to members in the distribution list. Users can enter this number in either the ShoreTel client or when addressing a message from the telephone user interface.

  This number cannot go beyond the range of numbers that you defined in the First System Distribution List Number and Last System Distribution List Number fields in the section from Chapter 2: Setting Up System Parameters starting on page 35. Also, the first system distribution list number is reserved for future use.

- **Recorded Name:** The buttons that correspond to this item let you Record, Play, Erase, or Import a recorded name for the distribution list.
  - — Click **Record** to record a name for the distribution list.
  - — Click **Play** to play back the recording.
  - — Click **Erase** to erase the recording.
  - — Click **Import** to import a sound file.
- **Language:** Select a language from the drop-down list.
- **Select from List: Filter Users By**—If you want to search for users with certain extension numbers or names beginning with a specific letter, use the Filter Users By fields. You can sort by Last Name, First Name, or Extension.

  If more names fit the criteria than can be displayed, use the Show Page controls. Either select a page number from the drop-down list or use the arrow keys to scroll through the list.

  After selecting a name (or names) from the user list, click Add to add it (or them) to the Distribution List Members box.

  If you want to remove a user from the directory list, select the user name from this box and click Remove.
- **AMIS Systems:** This lets you add users on AMIS systems to the distribution list. To add an AMIS user to the distribution list, select the AMIS system where the user you want to add is located and click Add. A dialog box prompts you for the extension number of the user. Enter the number and click OK. The AMIS System ID and extension (Mailbox ID) appears in the distribution list box.

13.2 AMIS Voice Mail

The ShoreTel system sends and receives voice mail messages to and from legacy voice mail systems by using Audio Messaging Interchange Specification (AMIS) protocol Version 1 — Spec February 1992. To send voice mail messages to remote AMIS sites, ShoreTel dials a phone number to access the remote system. Likewise, to receive voice messages from a remote system, the remote system must have the number to dial into the ShoreTel system. To reach the ShoreTel system, the remote system must be configured to dial a number that reaches an auto-attendant menu.

AMIS call support is enabled by default. Incoming AMIS voice mail is delivered in the same manner as other voice mail; however, replies cannot be sent. To send outbound AMIS voice mail, you must create AMIS systems in ShoreTel Director.

ShoreTel negotiates the setup, handshaking, and teardown of AMIS system calls. Each voice mail requires a call over the AMIS delivery and call-back numbers.

You can configure AMIS systems for two addressing methods. If the system does not use off-system extensions, a System ID number is required to direct the voice mail to the correct site. When a user wants to send a voice mail to a recipient on an AMIS system, he or she first must enter the System ID and then the mailbox number (extension).

Table 13-1 Examples of Address with a System ID

System ID	Recipient Mailbox Number
8331	1234
8408331	45657

If the system uses off-system extensions, these extensions become "off-system mailboxes." In this case, users simply address the voice mail by mailbox number and without entering the System ID.

Before you create AMIS systems to remote sites:

- Enable AMIS messaging from the Voice Mail Options page (default is enabled).
- Set permissions for the Voice Mail User Group to include dialing AMIS numbers. For more information, see the "Voice Mail Permissions" on page 323.
- Review the extension plans for all the systems to which you are making a connection. Make sure they use the same extension length and that extension numbers do not overlap.

After these global settings are done, creating AMIS systems require the following steps:

- Name the AMIS site and enter a System ID.
- Enter the phone number the ShoreTel system calls to connect to each remote AMIS system.
- Enter the phone number that remote AMIS systems call to send AMIS messages. This number must reach an auto-attendant.
- If a system is using off-system extensions, select the extension range for each AMIS system.

13.2.1 AMIS Restrictions

Some restrictions are placed on AMIS voice messages, as follows:

- ShoreTel establishes a call to an AMIS system for each voice mail. If a voice mail is addressed to multiple recipients, ShoreTel delivers as many as nine voice mails in a single call. If a voice mail has more than nine recipients, ShoreTel makes additional calls until the voice mail is delivered to all recipients. You can optimize AMIS voice mail delivery by using distribution lists at the remote AMIS sites.
- The maximum message length permitted is eight minutes.
- After ten failed attempts to complete a call to an AMIS system, ShoreTel disables the AMIS system and generates an event log.
- After ShoreTel establishes an AMIS system call, it tries three times to complete message delivery to each recipient. If ShoreTel fails to deliver a voice message after three attempts, it stops trying and returns the message to the sender. However, if the sender's voice mailbox is full, they do not receive failed messages.
- Outbound voice mail messages for disabled AMIS systems are accepted and queued. To deliver queued messages, enable the AMIS system in question from ***AMIS*** edit page (see Figure 13-3).

13.2.2 Enabling AMIS

To enable AMIS systems:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Voice Mail > Options**. The Voice Mail Options page shown in Figure 13-3 appears.

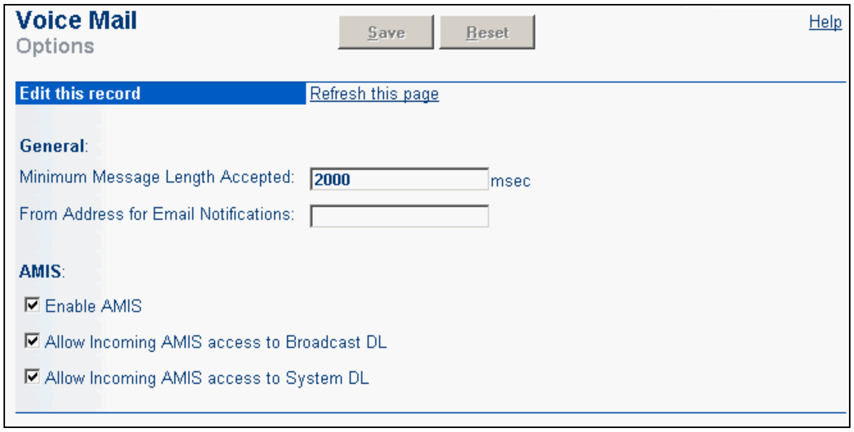


Figure 13-3 Voice Mail Options Edit Page

Step 3 Set the following set global AMIS parameters:

- **Minimum Message Length Accepted:** Specify minimum length a message must be to be acceptable. In milliseconds. The system default is 2000 milliseconds.
- **From Address for Email Notifications:** Specify the e-mail address to be placed in e-mail notifications about new messages.
- **Enable AMIS:** To enable AMIS systems support, click the check box (enabled by default). The Enable AMIS check box enables/disables all AMIS systems. Individual AMIS systems can be enabled and disabled from the AMIS edit page. For more information, see the "Creating AMIS Systems" on page 446.
- **Allow Incoming AMIS access to Broadcast DL:** To allow delivery of incoming AMIS messages to the Broadcast Distribution List, click the check box.
- **Allow Incoming AMIS access to System DL:** To allow delivery of incoming AMIS messages access to the System Distribution Lists, click the check box.

  The Voice Mail application will automatically remove silence from voice messages. If the resultant message after silence removal is less than this minimum message length, the message is assumed to be a hang-up and will be deleted from the system.

Step 4 Click **Save**.

13.2.3 Creating AMIS Systems

After you enable AMIS systems from the Voice Mail Options page, the next step is to create and configure the individual AMIS systems. Enter the AMIS delivery and call back number for each AMIS system you want to configure.

Expand the Voice Mail link in the navigation frame and click AMIS. The AMIS Systems list page appears as shown in Figure 13-4.

Figure 13-4 AMIS Systems List Page

To add a new AMIS system, click Add New. To edit an existing system, click an entry in the AMIS System column. The AMIS edit page appears, as shown in Figure 13-5

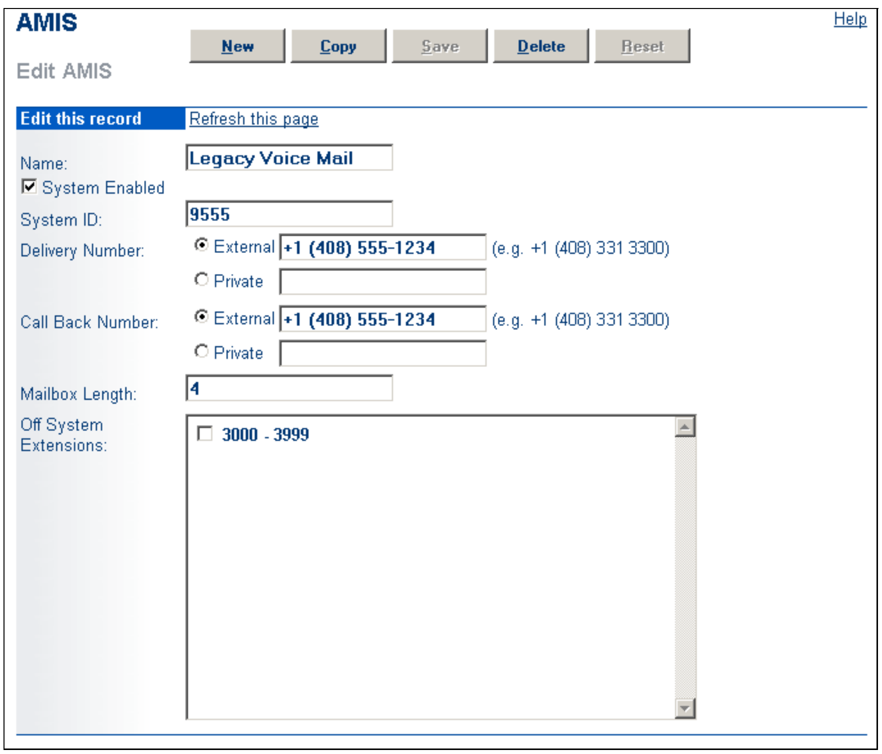


Figure 13-5 AMIS Edit Page

The following section describes the *AMIS* edit page parameters.

- **Name:** In the Name text box, enter the name of this AMIS site.
- **System Enabled:** Click this check box to enable this AMIS system.

  Outbound voice mail for this system is queued until the system is reset by clicking the System Enabled check box.

- **System ID**: The System ID defines the AMIS site where the voice mail for this system is delivered. The System ID plus a mailbox number identifies the site and the voice mail recipient. Plan the System ID to simplify the process of sending AMIS voice mail for your users. The System ID consists of an access code plus a site identifier.

  The System ID must begin with a digit reserved for trunk access codes, although it can be different from other trunk access codes. To make the System ID intuitive to voice mail users, choose a site identifier related to the public numbers used at the site.

  For example, if the voice mail delivery number is +1 (408) 555-1234, then System IDs like 8555 or 9408555 will be intuitive to your users. Generally, the shorter the System ID number, the easier it is to use.

  The System ID plus the mailbox length cannot exceed 15 digits.

  System IDs are required and can be single digits. Each AMIS system you create must have a unique System ID.

- **Delivery Number:** This is the number ShoreTel calls to send AMIS voice messages to the remote system. An external number is a public PSTN number and a private number is an internal, off-system extension connecting to an intra-site PBX system.

- **Call Back Number:** This is the number on which you receive AMIS messages. An external number is a public PSTN number and a private number is an internal, off-system extension connecting to an intra-site PBX system.

- **Mailbox Length**: Set the mailbox length of the remote sites mailboxes (extensions). If you are using off-system extensions, the length must match the length of your extensions.

  The System ID plus the mailbox length cannot exceed 12 digits.

- **Off-System Extensions:** If your system is using off-system extensions, select the extension range for each AMIS system. These extensions function as off-system mailboxes, allowing users to address voice mail to users on remote AMIS sites without entering a System ID. For more information, see the "Add or Edit a Trunk Group" on page 156.

13.2.4 Disabling AMIS Systems

You can disable AMIS systems globally or by individual connection. Individual AMIS systems are automatically disabled when ShoreTel fails to complete a call to an AMIS system.

To globally disable AMIS systems, deselect the Enable AMIS check box in the Voice Mail Options page (see Figure 13-3 on page 445). To disable an individual AMIS system, from the AMIS System list page, double-click the name of the AMIS system you want to disable. The AMIS edit page appears. Clear the System Enabled check box and click Sav*e*.

When you disable AMIS, ShoreTel will not send or receive AMIS voice messages.

Users can address outgoing voice messages while the system is disabled. Outbound messages are queued until the individual AMIS system is re-enabled. Attempts to deliver to a disabled AMIS system fail.

13.2.5 Setting Voice Mail User Group Permissions

You must set the permissions for the Voice Mail User Group to allow ShoreTel calls to the AMIS system delivery numbers you have configured. For instructions on setting these permissions, see the "User Groups" on page 325.

13.2.6 AMIS Test Mailbox

ShoreTel allows you to designate a mailbox that a remote AMIS system can use to test AMIS features. When you address a voice mail to the AMIS test mailbox, ShoreTel automatically replies with the same message.

Voice Mail reports are used by administrators to monitor, administer, and manage system resources and user activity.

13.3 Delivery and Notification

The user's voice mail notification options are edited from the Escalation Profiles and Other Mailbox Options page. Voice mail escalation parameters specify what should happen if notification is enabled at the time a message arrives. These options include the user's message notification telephone number, pager ID number, and try options.

Voice mail may be auto-forwarded. That is, a mailbox may be configured to send any message it receives to another mailbox. The message sent to the original mailbox can be automatically deleted, as an option. The target mailbox for forwarded messages may be any user, a workgroup, a route point, AMIS address, or a system distribution list (other than a broadcast distribution list). A message is pre-pended to the forwarded message, along with a time-stamp, announcing that the message has been auto-forwarded. As an example, the recipient of an auto-forwarded message might hear, "Auto-forwarded message received at 9:10 AM from Customer Support Mailbox".

An example of use might be the handling of off-hours calls when few support staff are available. Off-hours calls may be routed to a back-up extension. If no one is available to answer the back-up extension, calls may wind up in a voice mailbox that will not be checked for hours. The back-up extension can be set to auto-forward any calls that are received in its mailbox. Calls can be forwarded to a mailbox that is checked on a regular basis.

Auto-forwarding is available between distributed voice mail servers. The message is handled as any other message would be, including any message waiting indicator, any calling notification, return receipt requests, or urgent markings. Auto-forwarded messages can be forwarded and replied to. If the target mailbox is full, the is left in the sending mailbox. If a message is being auto-forwarded to a list of mailboxes and one is full, that target is skipped.

13.3.1 Escalation Profiles and Other Mailbox Options

The Escalation Profiles and Other Mailbox Options page can be accessed from a link on the User's Personal Options page. From here, you can configure Escalation Profiles to notify employees when a voice mail is received (which can be helpful in providing your customers with superior service/support after hours), and you can configure Automatic Message Forwarding, and Email Delivery Options for a user, so that users can be notified when their voice mailboxes are almost full.

The Escalation Profiles and Other Mailbox Options page is shown in Figure 13-6.

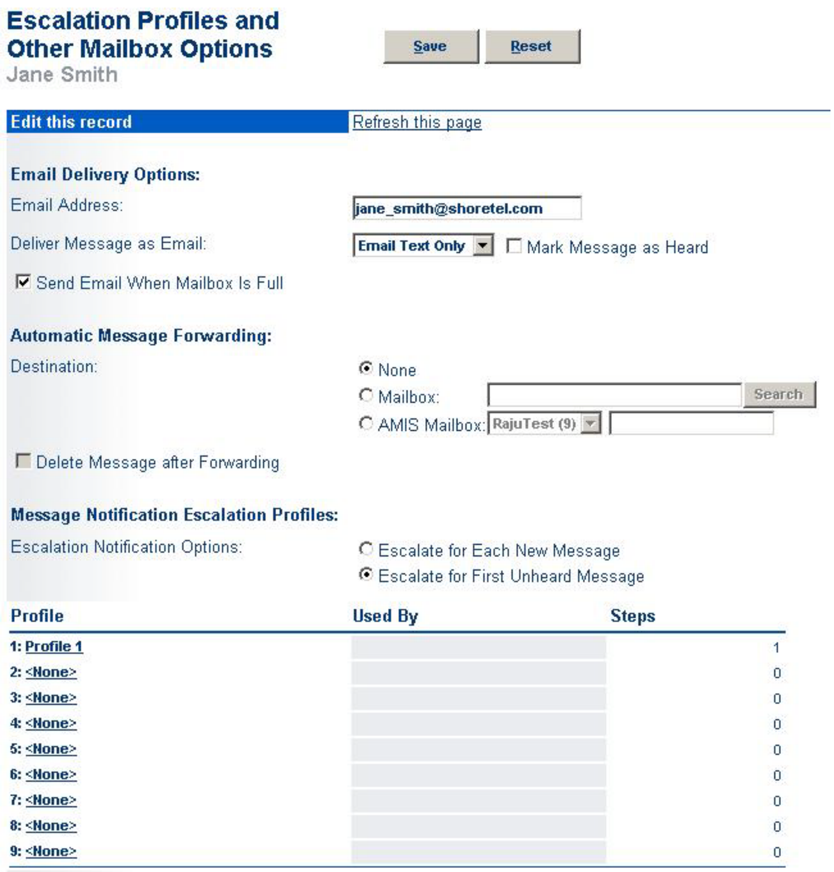


Figure 13-6 Escalation Profiles and Other Mailbox Options Page

13.3.1.1 Parameters

The Escalation Profiles and Other Mailbox Options parameters are as follows:

Email Delivery Options

- **Email Address:** This is the user's e-mail address. By default, it is automatically entered when you enter the user's first and last names in the First Name and Last Name fields. It consists of the first initial of the user's first name followed by the user's entire last name. In addition, the @*companyname*.com domain is saved in a cookie on the workstation each time you save a user. The system presents this information as a default, which can be changed as needed.

  Be sure to delete this field if the user does not have or use email.

- **Deliver Message as Email**: Select one of three e-mail delivery options:

  Disabled, Email Text Only, or Attach WAV File. The Email Text Only option notifies the user of the time, duration, and Caller ID of the message that was recorded. The Attach WAV File option attaches the voice message to the email as a WAV file.

  Selecting the Mark Message as Heard results escalation profiles being disabled for messages that are delivered to "generic" email address (i.e. not the ones in the escalation steps).

- **Send Email When Mailbox is Full:** Select the Send Email When Mailbox is Full check box to enable Voice Mailbox Full Notifications. This feature sends users a notice informing them that their mailbox is almost full. This message is sent when the user's mailbox approaches maximum capacity and crosses a non-configurable threshold (i.e. space for only 10 messages remaining).

See the "Configuring Voice Mailbox Full Notifications" on page 450 for more information.

Automatic Message Forwarding

- **Destination:** Messages may be auto-forwarded to a user, a workgroup, a route point, or a system distribution list. If the system distribution list includes AMIS destinations, they also receive the auto-forwarded message.

  The default is None, meaning, no forwarding.

  The destination may not be a broadcast distribution list.

- **Delete Message After Forwarding:** Select the checkbox to cause the automatic deletion of the message after forwarding. The default is not to delete.

Message Notification Escalation Profiles

- **Escalation Notification Options:** Escalation Paging is a traditional voice mail feature that allows support groups to offer round-the-clock service to their customers.

  Customers can call into your system when they have a problem, and can leave a voice message. This causes the Escalation Paging feature to begin notifying (via email and/or pager/voice message) the appropriate personnel.

  If the first person does not respond to the notification by listening to the customer's voice mail message within a certain time period, the next support person on the list is contacted, and so on, until as many as 10 people have been contacted.

  You can choose to begin the notification process for each new message that arrives in the voice mailbox by selecting the Escalate for Each New Message radio button, or you can select the Escalation Paging for First Unheard Message radio button so that the notification process will activate for only the first unheard voice message (in which case subsequent unheard messages would be regarded as redundant and would not trigger another wave of notifications).

  For more information on configuring this feature, see the "Configuring Escalation Notification" on page 452.

- **Profile:** Name of the Escalation Notification Profile.
- **Used By:** This is a user's Call Handling Mode (CHM) that is associated with the Escalation Notification Profile. A profile can be associated with one or more CHM's
- **Steps:** This is the number of steps in that profile.

13.3.1.2 Configuring Voice Mailbox Full Notifications

The Voice Mailbox Full Notification feature offers a way for users to receive an alert that tells them their mailbox is almost full before it gets to the point that they stop receiving messages.

When a user's mailbox approaches its maximum capacity (and a non-configurable threshold has been crossed), the system sends users a notice informing them that their mailbox is almost full and that there is only enough room for 10 additional messages. Each time users log into voice mail, they will receive a notice telling them how much space remains. In this way, mailbox owners are given adequate notice that they must clean up their mailboxes and they are not caught off-guard by an unexpected (and unwanted) “mailbox full” notification.

Recall that the maximum number of messages a user can receive ranges from 0 to 500 and can be set on the “Class of Service - Voice Mail” window in Director. This flexibility implies that not all users in the system will have the same upward limit to the number of voice mail messages they can receive.

Details:

- The mailbox warning threshold occurs when there is room for only 10 more messages in a user's mailbox. This threshold is non-configurable and is the same for all users, regardless of total mailbox capacity.
- As a user's mailbox approaches its limit, a warning message will be played indicating that the user has room only for “n” number of messages where the value “n” will be a countdown from 10 to 0. This message will be played when a user logs into the mailbox via the telephone user interface or Communicator.
- The “almost full” notification will be played until a users delete their messages, thereby reducing the number below the threshold.
- When a mailbox has finally reached its limit, the mailbox owner will be notified (if this option has been enabled for this user) and a warning NT event will be logged.
- When a message is deleted, it is no longer counted against the total capacity for a user's mailbox.
- Deleted messages are temporarily held in a “deleted messages” folder. Up to 200 deleted messages can be temporarily held. Once this limit is reached, the mailbox will be considered full and the user will be unable to receive new messages until the deleted messages have been purged. If this happens, the mailbox owner will receive a notification telling him, “Your mailbox is full. No more messages will be accepted until you purge your deleted messages.”
- Deleted messages can be manually purged by the user or automatically by the system. Automatic purging occurs on a nightly basis.

To enable the Voice Mailbox Full Notifications feature for a user:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Users > Individual User**. The Individual Users page appears.

Step 3 Select the user for whom you want to enable notification when their voice mail box is full. The Edit User page for that user appears.

Step 4 Click the **Personal Options** tab.

Step 5 Scroll to the bottom of the page and click **Escalation Profiles and Other Mailbox Options**. The Escalation Profiles and Other Mailbox Options for the user appears.

Step 6 Select the **Send Email When Mailbox is Full** check box.

Step 7 Click **Save** to store your changes.

Step 8 Repeat this process for each user for which you would like to configure mailbox full notifications.

13.3.1.3 Configuring Escalation Notification

The ShoreTel system supports Escalation Notification. This voice mail feature allows your organization to know when your customers need help.

For example, if a customer in a small town calls his local utility provider to complain about a power outage at 4 a.m., it is possible nobody would be in the office at that hour to handle his call. However, with ShoreTel's Escalation Notification feature, the customer could leave a voice mail, and in doing so, he would set in motion a chain of events that would cause support personnel from the utility company to respond to his concerns.

The message left by the customer on the voice mail system would trigger the Escalation Notification feature to send out a page, phone call, or email to an employee in the support department of the utility company. If this first employee ignores the beeping pager, another person will be contacted, and so on. Each of those utility company employees specified in the escalation profile will be contacted until someone dials into the ShoreTel system and listens to the customer's voice mail message and handles the problem. (See Figure 13-7 below.)

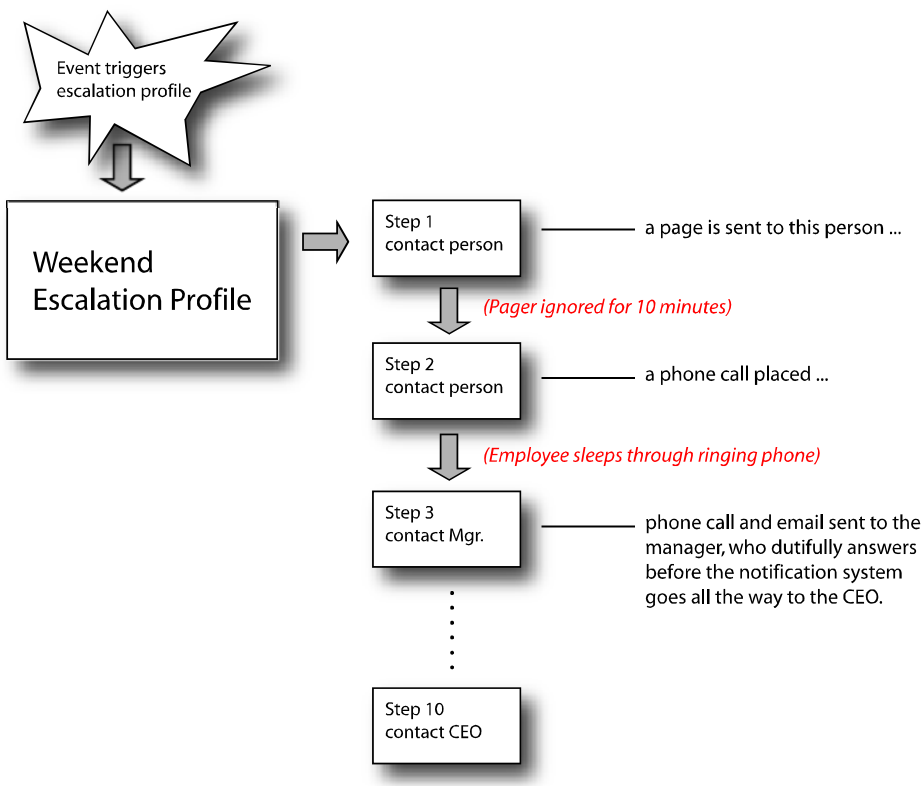


Figure 13-7 Anatomy of an Escalation Profile Event

Details for creating an Escalation Notification profile:

- Each escalation profile has ten notification steps, allowing the system administrator to specify who will be contacted at each step and the method used to contact that person (i.e. phone call or pager). An email can be sent to that person in addition to the phone call or pager notification.

- A maximum of nine notification profiles are supported.
- Call handling modes can be associated with different notification profiles.
- If a message is left, and someone listens to it, the notifications will stop. However, if someone marks a message unheard, that will restart the notification process in the same way that receiving a new voice message will.
- Escalation notification is supported on all mailboxes, including user mailboxes (extension and mailbox users, mailbox-only users, SMDI mailbox-only users) as well as workgroup mailboxes.

To configure Escalation Notifications via ShoreTel Director:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Users > Individual User**. The Individual Users page appears.

Step 3 Select the user for whom you want to enable notification when their voice mail box is full. The Edit User page for that user appears.

Step 4 Click the **Personal Options** tab.

Step 5 Scroll to the bottom of the page and click **Escalation Profiles and Other Mailbox Options**. The Escalation Profiles and Other Mailbox Options for the user appears.

Step 6 In the Message Notification Escalation Options section, do one of the following: select the appropriate radio button. Options are:

- Check the **Escalate for Each New Message** check box to have the system send a new wave of escalation notification messages each time a new voice mail message arrives. If several messages arrive within a short period of time, those who are notified will receive multiple notifications (when perhaps one notification would have been all that was required).
- Check the **Escalate for First Unheard Message** check box to have the system send an escalation notifications to begin at the receipt of the first voice mail message. Subsequent unheard voice mail messages will not trigger another wave of notifications as long as the first message remains unheard.

Step 7 In the Profile section, click the link for the desired escalation profile. The Escalation Profile page appears as shown in Figure 13-8.

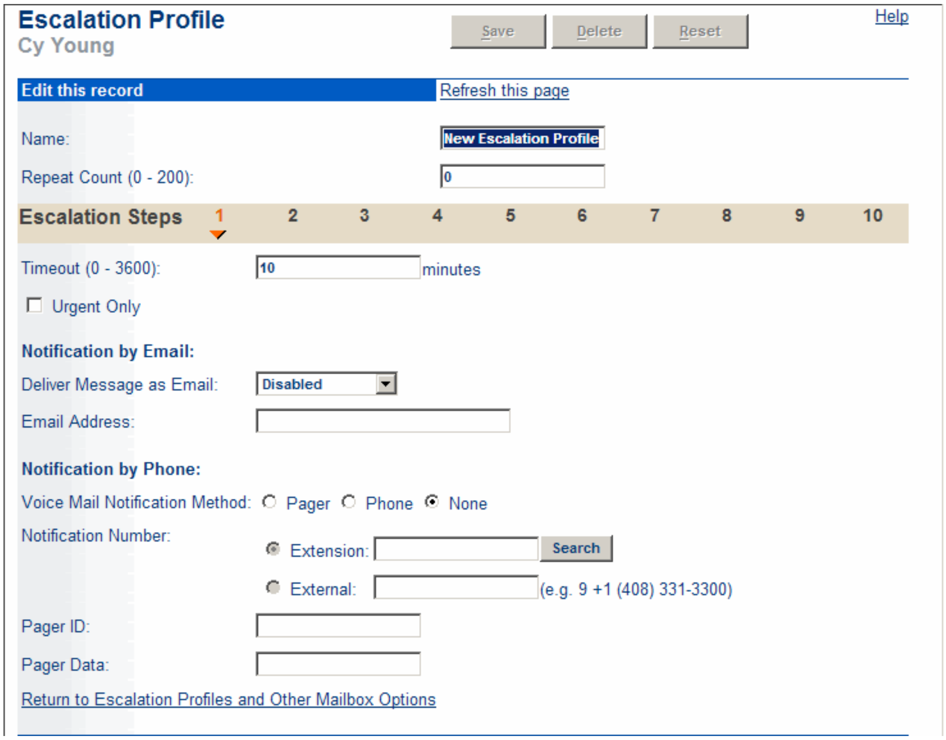


Figure 13-8 Escalation Profile Page

Step 8 Do the following to create an escalation profile:

Step a In the Name field, enter the name that you want to use for the profile.

Step b In the Repeat Count field, enter a value ranging from 0 to 200. This is the number of times the system will loop through the 10 steps of this profile before it quits trying to contact the various notification members. Selecting 0 will cause the escalation notification profile to execute once, without repeating. Selecting 1 will cause it to execute twice (i.e. once with one repeat loop).

Step c In the Timeout field, enter the number of minutes (0 - 3600) you want to elapse before the next step within this profile is executed. (This is the amount of time a message recipient has to respond to the original voice mail before escalation occurs.)

Step d Check the **Urgent Only** check box to have notification sent in only when the escalation is determined to be urgent.

Step e In the Deliver Message as Email field, select the method you want to use for email delivery. The options are as follows:

- Select **Disabled** to not send email notification.
- Select **Email text only** to have a text message sent to this user's email inbox. The message will contain basic information about the message (e.g. timestamp, sender, etc.)

- Select **Attach WAV file** to have a copy of the voice mail sent to the designated user's email inbox. This will allow the recipient to play the message from his or her PC.

Step f In the **Email Address** field, enter the email address of the first person that you want to notify.

Step g In the Voice Mail Notification Method section, select radio button for the method that you want to use to send voice mail notification: **Pager**, **Phone**, or **None**.

Step h In the Notification Number section, select radio button for the type of phone you want to send the voice mail message to, **Extension** or **External**, and enter the phone or pager number of the user in the field.

Step i In the Pager ID field, enter the pager pin number required to access the recipient.

Step j In the Pager Data field, enter the code the recipient requires to indicate that a page is waiting.

Step k Click **Save** to store your changes.

NOTE Click on the next Escalation Step (above the Timeout field) and repeat this process to configure up to ten steps within this escalation profile. Any unconfigured steps will be skipped over when the escalation profile is executed.

13.3.1.4 Linking an Escalation Notification Profile to a Call Handling Mode

To link an Escalation Notification Profile to a Call Handling Mode (CHM):

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Users > Individual User**. The Individual Users window appears.

Step 3 Select the user for whom you want to enable notification when their voice mail box is full. The Edit User page for that user appears.

Step 4 Click the Personal Options tab.

Step 5 Scroll to the Edit Call Handling Modes section and select the call handling mode that you want to associated with an escalation profile. The CHM page appears, as shown in Figure 13-9.

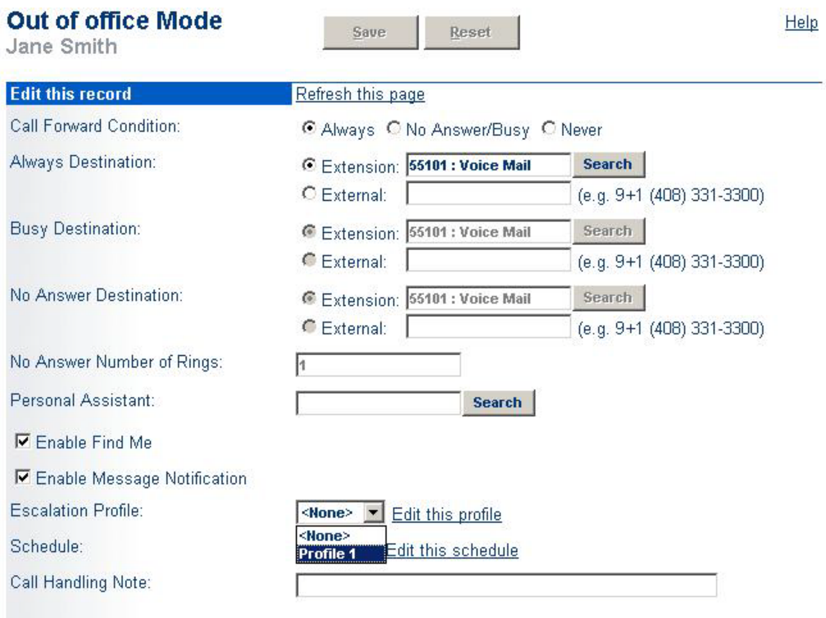


Figure 13-9 Selecting the escalation profile to associate with Out of Office CHM

Step 6 In the **Escalation Profile** field, select the desired escalation notification profile that you want to use.

Step 7 Click **Save** to store your changes.

Step 8 Repeat this process to associate different escalation notification profiles with each of the different Call Handling Modes as needed.

13.4 Voice Mail Reports

When an administrator requests a Voice Mail report, the system retrieves statistics from the main server and all distributed servers. ShoreTel provides two types of reports from this information:

- **Summary Report:** The Summary Report lists voice mail resource data for each application server on the ShoreTel network.
- **Server Report:** The User Report lists voice mail usage information for the specified application server and users assigned to the application server.

13.4.1 Summary Report

The Voice Mail Servers Maintenance Summary page displays the Voice Mail Summary Report. To open the Summary page, shown in Figure 13-10, select Maintenance -> Services -> Voice Mail from the Director menu.

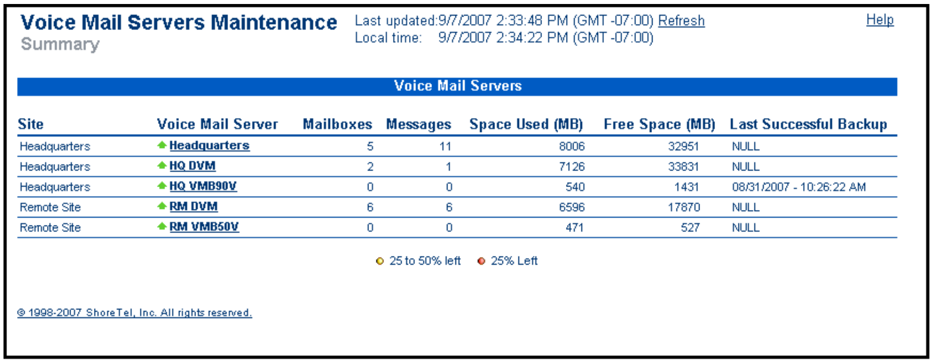


Figure 13-10 Voice Mail Statistics: Totals Report

The table lists all application servers configured on the network.Each row corresponds to one application server and each column lists a server property or resource statistic.

The Summary Report includes the following fields:

- **Site**: This parameter lists the site to which the server is assigned, as specified by the Edit Application Servers page.
- **Voice Mail Server**: This parameter lists the name of the server, as specified by the Edit Application Servers page.
- **Mailboxes**: This parameter lists the number of mailboxes on the server.
- **Messages**: This parameter lists the number of messages that are stored in the server's mailboxes.
- **Space Used (MB)**: This parameter lists the memory the server is using to store mailbox messages, user name recordings, auto attendant prompts, logs, and other data.
- **Free Space (MB)**: This parameter lists the hard disk memory available on the server.

  When less than 25% of memory capacity of the server is available, the page displays a red icon next to the free space value for that server. When the available memory is between 25% and 50% of capacity, the page displays a yellow icon next to the free space value.
- **Last Successful Backup**: This parameter lists the date that information on the server was backed up.

13.4.2 Server Report

The Voice Mail Servers Maintenance Server page displays a voice mail statistics report for the specified application server. To open the Summary page, shown in Figure 13-11, click the link of the desired application server on the Voice Mail Servers Maintenance Summary page.

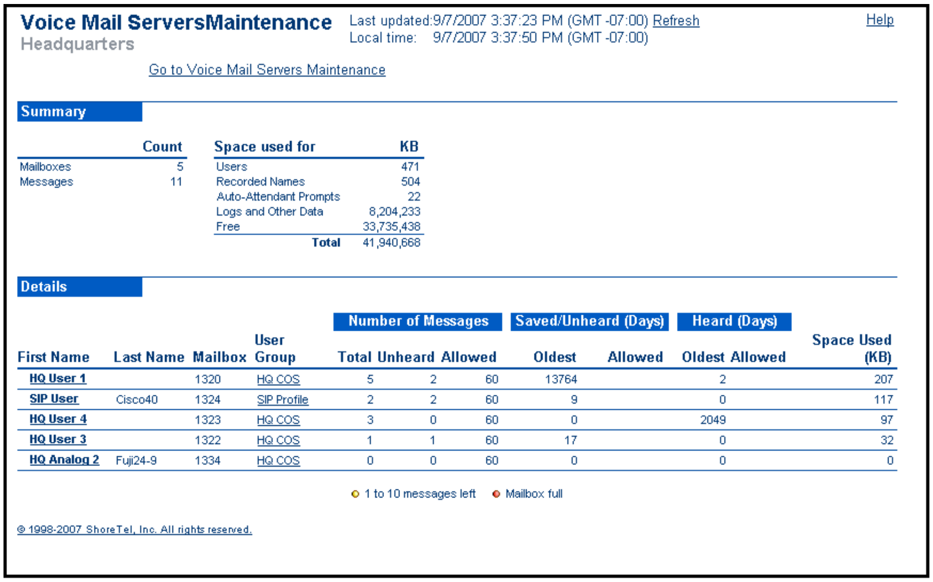


Figure 13-11 Voice Mail Statistics: User Report

The page lists the name of the application server for which the page is reporting statistics below the Voice Mail Servers Maintenance text in the upper left corner of the page. The page comprises two sections:

- **Summary:** The summary section lists resource usage and availability statistics for the specified application server.
- **Details:** The Details section lists resource usage statistics for users assigned to the application server.

13.4.2.1 Summary

The summary section, located on the top section of the Server page, lists resource usage and availability statistics for the specified application server. Statistics listed in this section include:

- **Mailboxes:** This parameter lists the number of mailboxes on the server.
- **Messages:** This parameter lists the number of messages that are stored in the server's mailboxes.
- **Space Used for:** These parameters indicate the manner in which the server's memory is allocated:
  - — **Users:** This parameter lists the memory used to store voice mail and mailbox configuration settings.
  - — **Recorded Names:** This parameter lists the memory used to store recordings of all user names.
  - — **Auto-Attendant Prompts:** This parameter lists the memory used to store auto-attendant prompts.
  - — **Logs and Other Data:** This parameter lists the memory used for log files generated by services hosted by the server.
  - — **Free:** This parameter lists the available unused memory resources.
  - — **Total:** This parameter lists the total memory resources on the server.

13.4.2.2 Details

The Details section lists resource usage statistics for the fifty largest mailboxes on the server. Mailboxes are sorted in order of the amount of disk space used to store their contents.

Each row corresponds to one mailbox on the application server. Each column lists a mailbox property or resource statistic.

The Details Report includes the following fields:

- **User Information:** These parameters identify the user to whom the mailbox is assigned. User information parameters are configured in the Edit User – General page.
  - **First Name:** This parameter lists the first name of the mailbox owner.
  - **Last Name:** This parameter lists the last name of the mailbox owner.
  - **Mailbox:** This parameter lists the extension of the mailbox.
  - **User Group:** This parameter lists the User Group to which the mailbox owner is assigned.
- **Number of Messages:** These parameters list the mailbox capacity and contents statistics.
  - **Total:** This parameter lists the number of messages in the mailbox.
  - **Unheard:** This parameter lists the number of messages that are marked as unheard.
  - **Allowed:** This parameter lists the capacity of the mailbox. The number of messages that a mailbox can hold is configured through Class of Service settings.
- **Saved / Unheard (Days):** These parameters list the age of the oldest message in the specified mailbox that has not been heard.
  - **Oldest:** This parameter lists the age, in days, of the oldest message that is marked unheard.
  - **Allowed:** This parameter lists when messages marked as unheard are removed from the server, in terms of the message age. This parameter is configured through Class of Service settings.
- **Heard (Days):** These parameters list the age of the oldest message in the specified mailbox that has been heard.
  - **Oldest:** This parameter lists the age, in days, of the oldest message that is marked heard.
  - **Allowed:** This parameter lists when messages marked as heard are removed from the server, in terms of the message age. This parameter is configured through Class of Service settings.
- **Space Used:** This parameter lists the memory required to store contents of the specified mailbox. This statistic includes the memory required to store messages that are deleted but not purged.

13.5 Voice Mail Synchronization with Gmail for Business

The Synchronization with Gmail for Business feature automatically synchronizes the state of a ShoreTel user's voice mail with the state of each corresponding email (when voicemail status is sent by way of ShoreTel's email notification). The HQ or DVS server monitors the state of a user's voice mails and emails and synchronizes those states. For example, when a user opens the voicemail notification email, the voicemail is marked heard on the voicemail system, and the message-waiting indicator on the phone is turned off.

NOTE In the current release, Google Gmail is the only email server that a ShoreTel system can interoperate with for synchronization of voicemail status. The feature currently works on Gmail Premier and Educational email accounts only. These accounts have the APIs that are necessary for the integration to work.

13.5.1 Background

A user account can be configured so that the user receives email notification of a new voice message. This email notification can also arrive with an attached WAV file of the actual voice message. The user can receive both the notification and the voicemail by way of the email client. However, in releases below Release 11.2, the status of the messages is not synchronized.

NOTE Below Release 11.2, email notification is not synchronized with the related voice message on the server: After a user listens to a message by playing the WAV file attached to the emailed notification, the message on the voicemail server remains marked as unheard. Even though a system administrator could specify in Director that messages be marked as heard after email notification is sent, this approach does not allow the ShoreTel system to determine whether the message has actually been read by the email client.

13.5.2 Voicemail Synchronization Terms

This section contains definitions of acronyms that are relevant to the feature.

- Gmail: Google's email service.
- IMAP4: Internet Message Access Protocol 4. An application-level Internet protocol used for accessing email from a remote mail server.
- OAuth: This open protocol supports secure API authorization from desktop and web apps through a simple, standard method. For more details, see Use of the OAuth Protocol section. To research OAuth, go to http://code.google.com/apis/accounts/docs/OAuth2.html.

13.5.3 Behavioral Details

This section describes a range of details about the feature. Some details are more visible, such as the consequences of user action on a message as described in the Synchronization Rules section. Certain internal and network-level details are also described.

13.5.3.1 General Details

This section contains some general technical details of the feature.

- To monitor the status of email, Synchronization with Gmail for Business uses the IMAP4 and OAuth protocols to access and authenticate with Gmail server. The Secure Sockets Layer (SSL) protocol is used to secure pertinent communications on the network.
- Access to user email by login with the username and password is not used.

13.5.3.2 Synchronization Service

The service runs on HQ and all DVS servers. The name of this service is ShoreTel-VmEmSync.

Each service is responsible for synchronizing the mailboxes that are on the same server. For example, the service running on HQ syncs the mailboxes located on HQ.

In the case of mailboxes on a Voice Mailbox Server switch (VMB switch), the synchronizing is done by the VmEmSync service that is running on the same server as the TMS that manages the VMB.

13.5.3.3 Synchronization Rules

The tables in this section contain synchronization rules. The applicable rules depend on whether an email notification has the attached WAV file and whether the server is synchronizing during ShoreTel-VmEmSync service startup or during normal operation.

13.5.3.4 Synchronization upon Startup

Start-up synchronization refers to initialization of the ShoreTel-VmEmSync service. This service gathers information on all voicemail and related email for each user and then synchronizes the states based on the rules in either Table 13-2 (for email text only) or Table 13-3 (for email with WAV file attachment). In these tables, the voicemail and email states occupy the two columns at left, and the resulting sync action is in the column at right.

Table 13-2 Start-up Synchronization with Email-only Text

Voice Mail State	Email State	Sync Action
Deleted	Not Deleted	Delete Email
Heard	Unread	Mark Email Read
All other states	Any state	No action

Table 13-3 Start-up Synchronization with WAV File Attachment

Voice Mail State	Email State	Sync Action
Heard	Unread	Email Read
Unheard	Read	Mark Voice Mail Heard
Deleted	Not Deleted	Delete Email
Not Deleted	Deleted	Delete Voice Mail

13.5.3.5 Synchronization during Normal Operation

Synchronization during normal operation is triggered when a user makes a change to a voicemail or email. In Table 13-4 and Table 13-5, the voicemail change is in the left column, and the consequence for the voicemail or email is in the right column.

Table 13-4 Sync with Email-only Text during Normal Operation

Event	Sync Action
Voicemail is deleted.	Delete email.
Voicemail is heard	Mark email as read.
All other events.	No sync action.

Table 13-5 Sync with WAV File Attachment during Normal Operation

Event	Sync Action
Voicemail is deleted.	Delete email.
Voicemail is heard.	Mark email as read.
Voicemail is undeleted.	Move email to Inbox. Mark email as unread if voice mail is unheard.
Voicemail is marked unheard.	Mark email as unread if voicemail is in “NEW” folder.
Email is deleted.	Delete voicemail.
Email is read.	Mark voicemail as heard.
Email is undeleted.	Move voicemail to “Saved” folder.
Email is marked unread.	Mark voicemail as unheard if email is not in “Trash” folder.

13.5.3.6 Usage of Network Resources

Before setting up this feature, consider its use of network resources as described in this section.

The ShoreTel system monitors the state of all user messages so that, for example, when a voice message is heard, the system reflects the state change in a timely manner. To ensure timely updates to the status of all messages, the system uses network bandwidth in proportion to the number of messages.

For example, consider a ShoreTel deployment that supports 1000 users and that each user has 5 messages. The state of 5000 messages total is monitored by the ShoreTel system. For monitoring the state of 5000 messages, the required bandwidth is 75 Kbytes per second. In this scenario, the time to synchronize a message’s state change between voice mail and email is less than 20 seconds.

In the event of a server restart, the initial synchronization time for a system with up to 1000 users is less than 3 minutes.

13.5.3.7 Synchronization Criteria

Synchronization is automatically enabled for a user if both of the following are true:

1. The user is configured to receive email notifications. The email address must be that of the user’s Premier/Education Gmail account.

2. The system administrator configured an email server with the domain for the user's email address (by using OAuth consumer key and secret strings). As an example, the system administrator configured OAuth access with the domain for the user's email address (explained further in 'Google OAuth Configuration' and 'ShoreTel Director Configuration' sections).

13.5.3.8 Use of the OAuth Protocol

The OAuth protocol lets a 3rd party gain access to a user's account without needing the user's password. By relying on OAuth, the ShoreTel-VmEmSync service can use the IMAP4 AUTHENTICATE command to examine a user's email (without logging in as the user). Gmail and most Google APIs support OAuth.

For the Premier and Education versions of Gmail, OAuth is set up by the system administrator. The admin enables certain capabilities and acquires a system-generated consumer secret at the Google OAuth management web page. The system administrator must first perform these actions in the applicable Google page before providing access to all accounts on a domain. An example of the Google Apps web page appears in Figure 13-12. In Figure 13-12, the machine-generated "OAuth consumer secret" is the value that the admin copies to a new Gmail configuration area of Director (described in Implementation).

The consumer secret from Google OAuth management and the consumer key (also described in Implementation) allow the ShoreTel HQ server or DVS to:

- Authenticate with Google mail servers without needing the user passwords.
- Establish a trusted host relationship between the two servers.

NOTE The Google Apps web page can change without notice, so the admin might see a variation of what appears in Figure 13-12.

13.5.4 Configuration

This section describes how to set up this feature but first provides some prerequisite information.

ShoreTel synchronization with Gmail Premier and Education Services utilizes a Google Apps OAuth key and user secret. The origins of the key and secret are described in the Google OAuth Configuration and ShoreTel Director Configuration sections.

13.5.4.1 Google OAuth Configuration

This section describes the steps that a system administrator performs at the Google Apps web site (for OAuth Management) before performing the steps in ShoreTel Director.

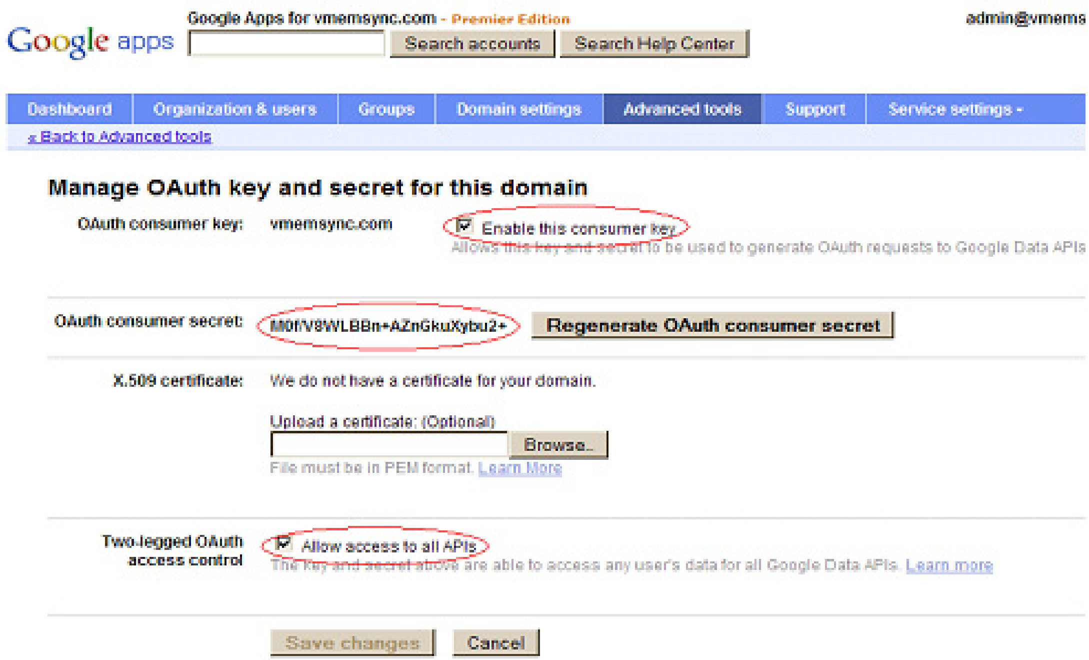


Figure 13-12 Google OAuth Management Page

To do the requisite steps at the Google Apps site:

Step 1 Navigate to the current Google Apps page (see example page in Figure 13-12).

Step 2 Put a check in the following two checkbox enables:

- "OAuth consumer secret . . . Enable this consumer key"
- "Two-legged OAuth access control . . . Allow access to all APIs"

Step 3 At "OAuth consumer secret," click the button labeled Regenerate OAuth consumer secret. A secret is generated and displayed.

Step 4 Write down or otherwise copy the secret. This secret must be entered in ShoreTel Director as described in ShoreTel Director Configuration.\

This completes the needed steps at the Google Apps site.

13.5.4.2 ShoreTel Director Configuration

An existing Director page has a new area for setting up Synchronization with Gmail for Business. The page is "System Parameters – Edit Other Parameters," and the new area is labeled Gmail Configuration, as Figure 13-13 shows.

Figure 13-13 New System Parameters - Edit Other Parameters Fields

The consumer key is what is listed on Google's OAuth Management Page. It is visible in Figure 13-12 on page 464, next to the label "OAuth consumer key:"

The consumer secret comes from the Google Apps domain management web site.

To activate the feature in Director:

Step 1 Open Edit Other Parameters by navigating to Administration –> System Parameters –> Other.

Step 2 The "OAuth consumer key" from the Google OAuth management page should be entered in the box labeled Gmail Consumer Key.

Step 3 Type the secret that was generated in the Google Apps web page in the box labeled Gmail Consumer Secret.

Click the Save button at the top of the Edit Other Parameters window.

CHAPTER 14

Configuring the Auto-Attendant

This chapter describes how to configure Auto Attendant.The topics include:

An Auto Attendant is a program that answers and handles inbound calls without human intervention. Auto attendants typically provide menu-driven options through which callers can obtain information, perform tasks, or connect to a requested extension.

The auto-attendant can answer incoming calls and transfer a caller to an extension, a mailbox, another menu, a workgroup, or a route point. It also includes a dial-by-name feature that transfers callers to the system directory, where they can connect to an extension by dialing the user's name.

14.1 Multiple Auto-Attendants

Multiple auto-attendants can be configured for different user groups or departments, and each auto-attendant configuration can have multiple levels of menu options.

There are no hard limits to the number of Auto-Attendants that can be configured in a ShoreTel system. However, in most installations, the system can support up to 500 AA menus. This number may be affected by the complexity of your dialing plan.

When the “main” auto-attendant is reached, it provides options for forwarding calls to individual user extensions. It can also provide options for forwarding calls to the sales department and customer operations department auto-attendants. From the sales or customer operations auto-attendants, callers are given options that transfer calls to the appropriate extension.

The dial-by-name operation of the auto-attendant can be limited to a department or other organizational sub-group by associating the operation with an extension list. To create extension lists, see the "Extension Lists" on page 358. Only users that have been selected to be included in the dial-by-name list will be included. For more information, see the "Individual Users" on page 329.

When callers are transferred back to the auto-attendant, either willingly or because of an error, they are returned to the default auto-attendant menu on the associated server.

14.2 Menus

The auto-attendant *Menus* page is where you begin the configuration to add a new auto-attendant menu or edit an existing menu.

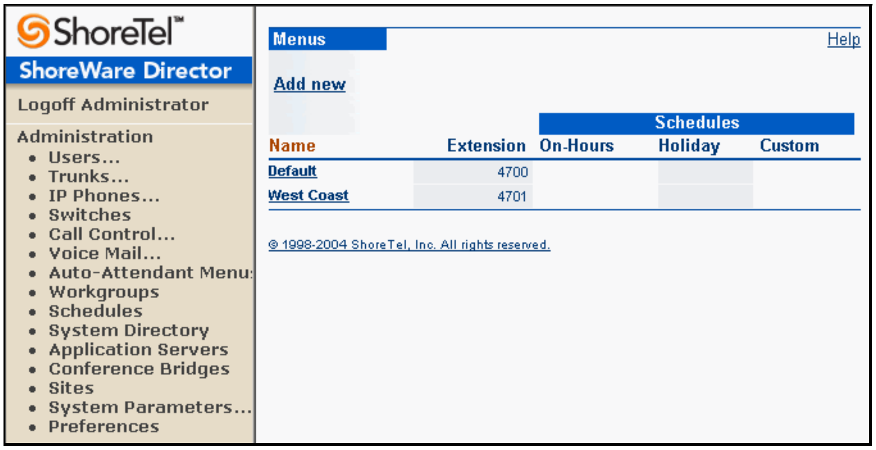


Figure 14-1 Auto-Attendant Menus List Page

14.2.1 Parameters

The parameters that appear on the *Menus* page are as follows:

- **Name:** This is the name of an existing auto-attendant menu configuration. Clicking an auto-attendant invokes the ***Menus*** edit page.
- **Extension:** This is the extension that is associated with an existing auto-attendant menu.
- **On-Hours:** This is the name of the On-Hours schedule, if any, that is associated with an existing auto-attendant menu.
- **Holiday:** This is the name of the Holiday schedule, if any, that is associated with an existing auto-attendant menu named in the Name column.
- **Custom:** This is the name of the Custom schedule, if any, that is associated with an existing auto-attendant menu.

14.2.2 Adding and Editing an Auto-Attendant Menu

To add a new or edit an existing auto-attendant menu, invoke the Menus page. From the Menus page, click Add new, or click an existing menu name in the Name column. This invokes the Edit Menu page, as shown in Figure 14-2.

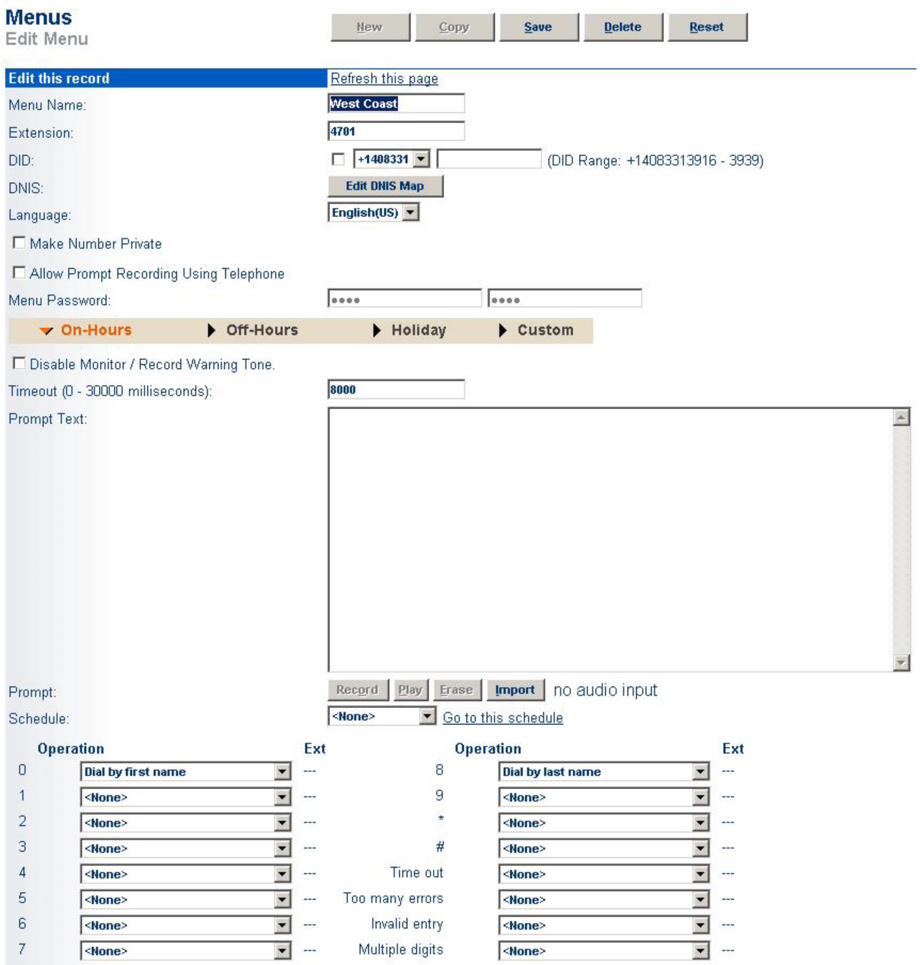


Figure 14-2 Edit Menu Page

14.2.2.1 Parameters

The parameters that appear on the Menu edit page are as follows:

- **Menu Name:** This is the name of the auto-attendant menu.
- **Extension:** This is the extension number associated with the auto-attendant menu. It must fall between the first and last menu numbers defined on the Dialing Plan edit page under System Parameters. See the "Setting Dial Plan Parameters" on page 35.
- **DID:** When the check box is selected, a DID number is used to access the associated auto-attendant menu. If you are adding a new menu or editing an existing menu and you want to access it using a DID number, check this box and enter a DID number in the accompanying text-entry field.

  Refer to "Assigning DID Numbers from a Range" on page 165 for additional information about DID numbers and ranges.
- **DNIS:** Click this link to set up one or more DNIS mappings to this menu. For more information about DNIS, refer to "Edit DNIS Digit Map" on page 167.

- **Language:** Select a language from the drop-down list. This is the language that will be used by the auto-attendant menu for responses such as "invalid entry". Greetings must be recorded in this language.
- **Make Number Private:** Checking this box removes this number from the system directory and call handling destination lists.
- **Allow Prompt Recording Using Telephone:** Select this check box to enable the User Recording of Auto-Attendant Menus via the menu mailbox.

  User recording of auto-attendant menus allows end users to dial into the system to record AA prompts in the same way that they would change their personal mailbox greeting (i.e. without having to access the recording interface through ShoreTel Director). This frees the system administrator from the task of recording AA menus, allowing him or her to delegate the task to more appropriate team members.
- **Menu Password:** A separate "Menu Mailbox" is created for each AA menu, allowing users to dial into the system to change the menu prompts. Each AA menu may have its own password and a unique, dialable number.

  If a password is desired, enter the password in the field provided, and enter it a second time to confirm.
- **On-Hours / Off-Hours / Holiday / Custom:** These are the operating modes for a new or existing auto-attendant menu. You can configure them for different situations. Click the appropriate link to view the schedules. Schedules are set using the Schedules link. See Chapter 15: Configuring Schedules starting on page 475.
  - **On-Hours** mode lets you configure the auto-attendant to handle incoming calls during regular office hours.
  - **Off-Hours** mode covers all hours not scheduled in other modes. This is typically when the office is closed for the evening and weekend.
  - **Holiday** mode lets you configure how the auto-attendant functions on holidays.
  - **Custom** mode is used for single days that are not covered by the other modes such as a company special event.
- **Disable Monitor/Recording Warning Tone:** This check box can be used to stop playing the warning tone for call recording and monitoring if the tone is turned on in the Call Control page. To see where the warning tone is first set, refer to the "Call Control Options" on page 290.

WARNING ShoreTel, Inc. does not warrant or represent that your use of call monitoring or recording features of the Software will be in compliance with local, state, federal or international laws that you may be subject to. ShoreTel, Inc. is not responsible for ensuring your compliance with all applicable laws.

Before disabling the warning tone, you may wish to consult with legal counsel regarding your intended use.

- **Timeout:** Set a timeout between 0-30000 milliseconds. This is the time the caller has to perform an action.
- **Prompt Text:** Before recording a prompt for a new or existing auto-attendant menu, enter the text for the prompt in this field. This also provides a convenient record of your prompt if you should ever need to re-record the prompt.

  This is an optional parameter.

Prompts on the ShoreTel system can be imported into the system using µ-law, WAV file format. If you would like your prompts to match the voice of the ShoreTel system, please contact Worldly Voices at www.worldlyvoices.com and request that "Connie" record your prompts. Worldly Voices provides this service with a rapid turnaround time for a nominal fee.

- **Prompt:** The Record, Play, Erase, and Import buttons associated with this parameter are used to record your auto-attendant menu prompt.

  Click Record to record the prompt; Play to play it back; Erase to erase the prompt; and Import to import a prerecorded prompt from a sound file.

- **Schedule:** This drop-down list shows the auto-attendant schedules: On-Hours, Off-Hours, Holiday, and Custom. Select a schedule from this list and click Go to this schedule to invoke the schedule so that you can configure a new or existing auto-attendant menu's schedule.

  See Chapter 15: Configuring Schedules starting on page 475, for information about setting up schedules.

- **Operation:** Each item in the Operation drop-down list lets you select the action that is associated with its dialpad number. This number is located to the left of each Operation drop-down list. When prompted by the auto-attendant, the caller is asked to enter this number.

  — **Dial by first name** lets the caller spell the user's first name from the dialpad. The auto-attendant then transfers the caller to the user's extension. To limit the dial list to a department or other organizational sub-group, select an extension list from the ***Ext*** column.

  — **Dial by last name** lets the caller spell the user's last name from the dialpad. The auto-attendant then transfers the caller to the user's extension.

  To limit the dial list to a department or other organizational sub-group, select an extension list from the Ext column.

  — **Go to extension** lets the user enter the extension he needs. This functions the same as a transfer but without a voice prompt.

  — **Go to menu** transfers the caller directly to the user's mailbox without ringing the user's extension. This is also used to send the caller to another menu. You must select the destination from the extension (Ext) pop-up dialog box.

  — ***Hang up*** lets the caller disconnect the call.

  — **Repeat prompt** lets the user hear the prompt again.

  — **Take a message** lets the caller leave a message by selecting a user's extension.

  — **Take a message by first name** lets the caller leave a message by selecting a user's name from the menu.

  — **Take a message by last name** lets the caller leave a message by selecting a user's name from the menu.

  — **Transfer to extension** transfers the caller to the user's extension where he or she can speak with the user or leave a message if the user does not answer. You must select a destination from the extension pop-up dialog box.

  Dial by last name is supported by default.

- **Ext:** This pop-up dialog box lets you select the destination that is associated with the Go to Menu or Transfer to extension operation.

Used this field to select an extension list to be used by the Dial by First Name and Dial by last Name operations.

- **Time Out:** This drop-down list lets you specify the action that the auto-attendant takes when the caller does not press a dialpad key in a system-defined period of time. Typically, the action is Repeat Prompt.
- **Too Many Errors:** This drop-down list lets you specify the action that the auto-attendant takes when the caller presses an invalid key too many times in a row. You might specify a user extension, such as the operator, for this. Typically, the action is Hang Up. If no action is specified, Hang Up is invoked by default.
- **Invalid Entry:** This drop-down list selects an action to take when a key is pressed that the auto-attendant does not recognize. Typically, the action is Repeat Prompt.
- **Multiple-Digit:** This drop-down list lets you select a multiple-digit action that the caller takes. The choices are None, Transfer to Extension, Take a Message, Go to Extension, and Go to Menu. The default is None.
  - **None** assigns no multiple-digit operation to the menu.
  - **Transfer to Extension** assigns a multiple-digit operation to the menu and prompts the caller to dial directly into a user's extension.
  - **Go to Menu** assigns a multiple-digit operation to the menu and prompts the caller to dial directly into a user's mailbox.

14.2.3 Configuring an Auto-Attendant Menu

To configure an auto-attendant menu from the ***Menus*** page:

Step 1 Enter the name of the menu in the Menu Name field.

Step 2 If this is a new menu, enter the menu's extension in the Number field. If you are editing an existing menu, enter a new extension in this field if necessary.

Step 3 If the menu will be associated with a DID number, check the DID check box and enter the DID number in the DID text-entry field.

Step 4 To associate the menu with a DNIS number, click **Edit DNIS Map** and set the map.

Step 5 Make Number Private to remove the number from the system directory.

Step 6 Select the **Allow Prompt Recording Using Telephone** check box to enable the User Recording of AA menus, and enter and confirm a password for the associated mailbox.

Step 7 Click the auto-attendant mode—On-Hours, Off-Hours, Holiday, or Custom—that to associate with the menu.

Step 8 Set a Timeout.

Step 9 Enter the text that you will use for recording the menu's prompt in the Prompt Text field. This is optional.

Step 10 Do one of the following:

- Click **Record** to record the prompt.

- Click **Play** to hear the prompt.
- Click **Erase** to erase it.
- Click **Import** and select the file from the appropriate directory to import a prerecorded prompt from a WAV file.

Step 11 Select a schedule for this menu from the Schedule drop-down list, and click **Go to this schedule**. See Chapter 15: Configuring Schedules starting on page 475 for information about schedules.

Step 12 Select the action that the auto-attendant takes in response to each supported digit from the Operation drop-down list.

Step 13 Assign an extension to the Operation (if applicable) from the extension (Ext) pop-up dialog box.

Step 14 Select an action from the Time out drop-down list, and select an extension from its extension pop-up dialog (if applicable).

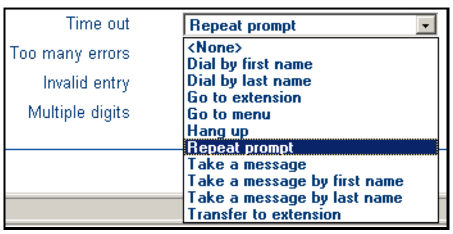


Figure 14-3 Timeout Errors Drop-Down List

Step 15 Select an action from the *T*oo many errors drop-down list and select an extension from its extension pop-up dialog box (if applicable).

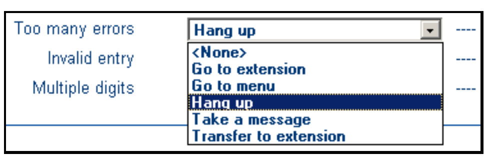


Figure 14-4 Too Many Errors Drop-Down List

Step 16 Select an action from the Invalid Entry drop-down list, and select an extension from its extension pop-up dialog box (if applicable). The options are the same as are available in the Timeout pop-up.

Step 17 Select an action from the Multiple digits drop-down list.

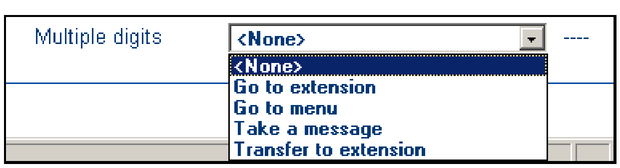


Figure 14-5 Multiple Digits drop-Down List

Step 18 Click Save to save the configuration.

14.2.3.1 Configuring Multiple Auto-Attendant Menus

Configuring multiple auto-attendant menus lets you add menus to your main menu so that callers can be directed to other departments in your company.

To add multiple auto-attendant menus to your main menu:

Step 1 Go to the Menus page and configure the menu you are adding to the main auto-attendant menu, as described in the "Adding and Editing an Auto-Attendant Menu" on page 468.

Step 2 Go back to the main auto-attendant menu's configuration on the Menu edit page.

Step 3 Go to the Operations section on the Menu edit page and select **Go to menu**.

Step 4 Associate the menu you are adding with a digit and the menu's extension. For example, if the menu's extension is 503, select this number from the **Ext.** drop-down list.

Step 5 Click **Save** so that your changes are recorded.

You can define the auto-attendant's schedule when configuring an auto-attendant menu from the Menus page or by invoking the Schedules link. Refer to Chapter 15 for information about establishing schedules.

The following schedule pages can be edited from the Menus page or from the Schedules link:

- On-Hours
- Holiday
- Custom

With the exception of the Off-Hours mode, each mode has a schedule configuration page. Off-hours is equal to all time not entered in the other schedules.

The following logic determines which schedule is active:

1. The auto-attendant first looks for the Custom schedule.
2. If the Custom schedule is not available, the auto-attendant looks for the Holiday schedule.
3. If the Custom or Holiday schedule is not available, the auto-attendant looks for the On-Hours schedule.
4. If the Custom, Holiday, or On-Hours schedule is not available, the auto-attendant looks for the Off-Hours schedule.

ShoreTel Director forms the Off-Hours schedule from all the hours not scheduled in the other modes. If you do not create a schedule for at least one of the other modes, the Off-Hours schedule will be all possible hours.

CHAPTER 15

Configuring Schedules

This chapter describes how to create schedules for the ShoreTel system. The topics discussed include:

Schedules let you to define business hours and can facilitate proper routing of inbound calls. Schedules can be used by Hunt Groups and by the Auto-Attendant.

The ShoreTel system supports the following types of schedules:

- On-Hours
- Holiday
- Custom

Hours for on-hour and custom schedules are configurable. Holiday schedules let you identify the days when your organization is otherwise not open for business. Off-hours are considered all time that is not entered in the other schedules.

The following logic determines which schedule is active:

1. The auto-attendant or hunt group first looks for the Custom schedule.
2. If the Custom schedule is not available, the auto-attendant or hunt group looks for the Holiday schedule.
3. If the Custom or Holiday schedule is not available, the auto-attendant or hunt group looks for the On-Hours schedule.
4. If the Custom, Holiday, or On-Hours schedule is not available, the auto-attendant or hunt group looks for the Off-Hours schedule.

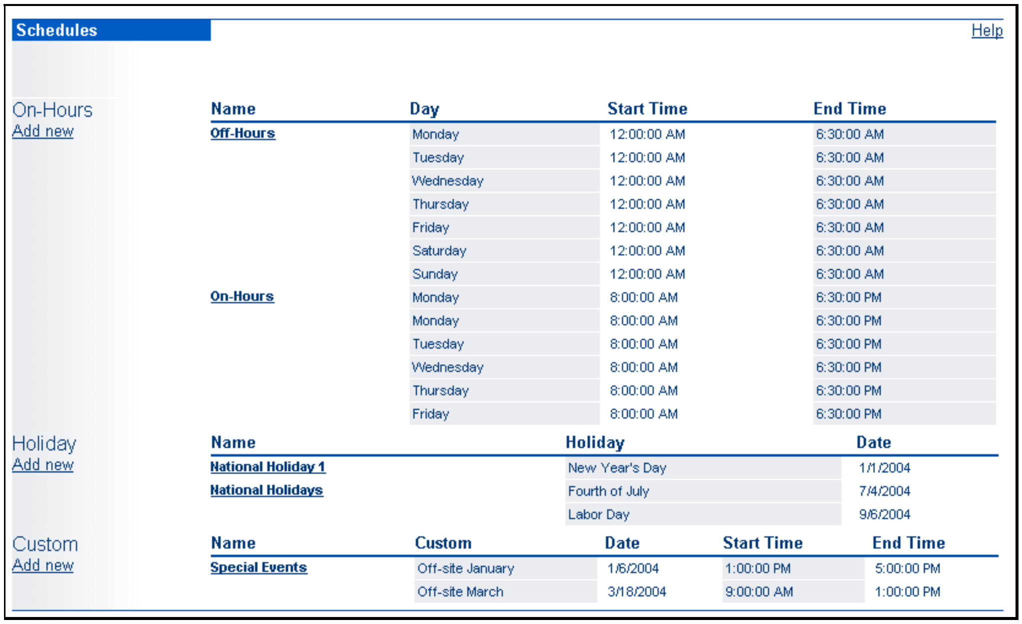
ShoreTel Director forms the Off-Hours schedule from all the hours not scheduled in the other modes. If you do not create a schedule for at least one of the other modes, the Off-Hours schedule will be all possible hours.

15.1 Accessing the Scheduling Page

To access the scheduling page, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Schedules**. The Schedules page appears as shown in Figure 15-1.



Schedules | Help

On-Hours | Add new

Name	Day	Start Time	End Time
Off-Hours	Monday	12:00:00 AM	6:30:00 AM
	Tuesday	12:00:00 AM	6:30:00 AM
	Wednesday	12:00:00 AM	6:30:00 AM
	Thursday	12:00:00 AM	6:30:00 AM
	Friday	12:00:00 AM	6:30:00 AM
	Saturday	12:00:00 AM	6:30:00 AM
	Sunday	12:00:00 AM	6:30:00 AM
On-Hours	Monday	8:00:00 AM	6:30:00 PM
	Monday	8:00:00 AM	6:30:00 PM
	Tuesday	8:00:00 AM	6:30:00 PM
	Wednesday	8:00:00 AM	6:30:00 PM
	Thursday	8:00:00 AM	6:30:00 PM
	Friday	8:00:00 AM	6:30:00 PM

Holiday | Add new

Name	Holiday	Date
National Holiday 1	New Year's Day	1/1/2004
National Holidays	Fourth of July	7/4/2004
	Labor Day	9/6/2004

Custom | Add new

Name	Custom	Date	Start Time	End Time
Special Events	Off-site January	1/6/2004	1:00:00 PM	5:00:00 PM
	Off-site March	3/18/2004	9:00:00 AM	1:00:00 PM

Figure 15-1 Schedules Pane

15.2 Configuring the On-Hours Schedule

To configure the On-Hours schedule, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Schedules**. The Schedules page appears.

Step 3 In On-Hours section in the Name column, click **On-Hours** to modify the existing schedule. You can also click Add new under the On-Hours heading to create a new on-hours schedule. The Edit On-Hours Schedule page appears as shown in Figure 15-2.

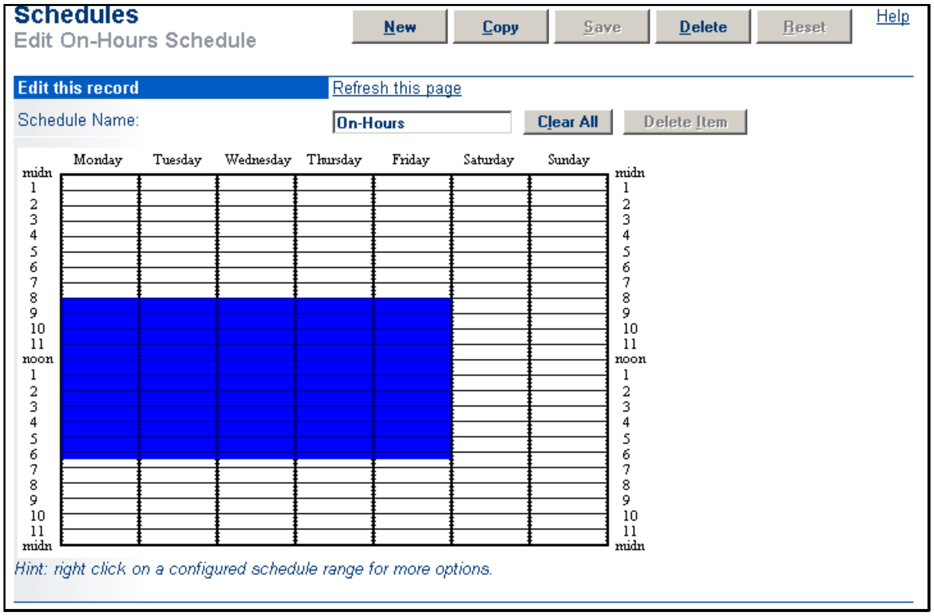


Figure 15-2 On-Hours Schedule Edit Page

Step 4 Create hour blocks by clicking in the hour slot for the start-time of a work day and drag the mouse to the end-time for block that day. Right-click to save.

Step 5 Repeat Step 4 for each workday. You can modify time frames on the fly by clicking the right mouse button to invoke the options shown in Figure 15-3.

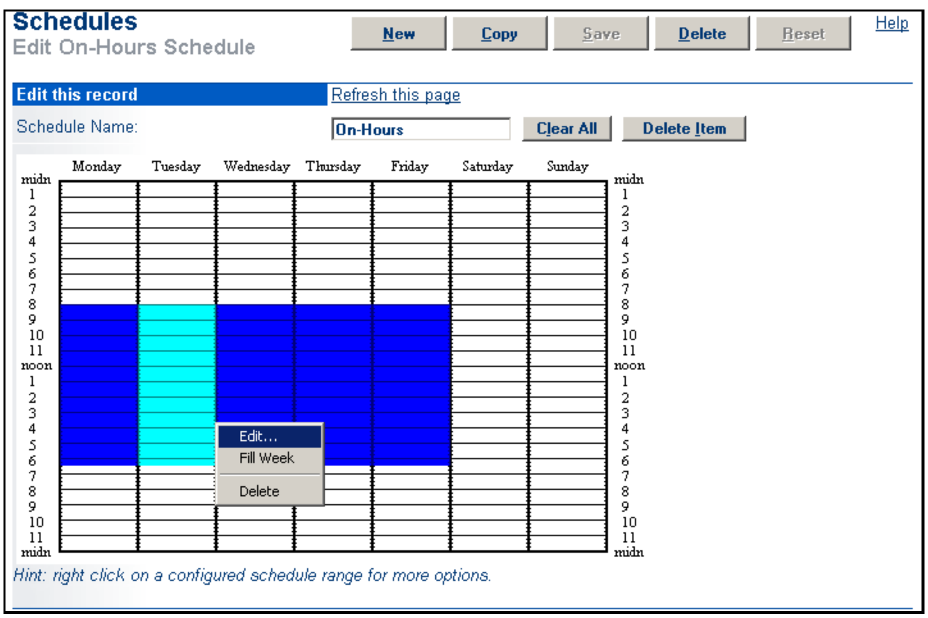


Figure 15-3 Time Frame Options

The options include:

- Click **Fill Week** to populate the other days of the week with the same schedule.
- Click **Edit** to edit the selected time block.
- Click **Delete** to remove the selected time block.

If you want to change the entire schedule, you must delete all entries in the schedule; otherwise, duplicate entries will be made.

Step 6 Click **Save** to save the schedule in the database.

You can schedule a mode to start and stop multiple times in one day. For example, on Monday, you can set the schedule to start at 4:30 am and stop at 9:00 am, and then schedule it to resume at 2:00 pm until 5:30 pm by performing Step 4 and Step 5 again.

15.3 Holiday Schedule

The Holiday Schedule edit page is shown in Figure 15-4.

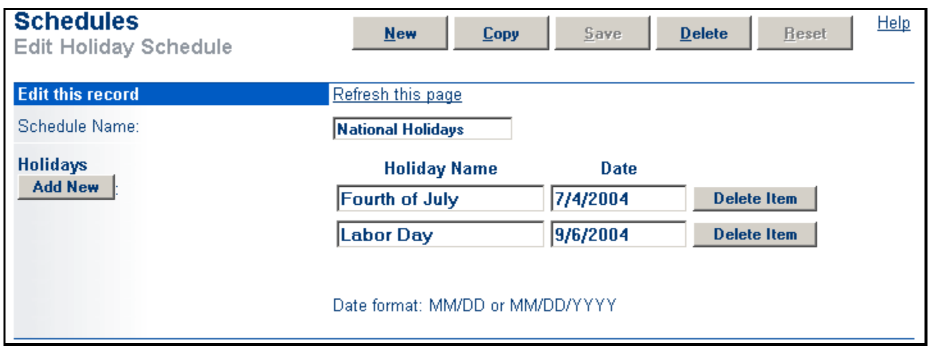


Figure 15-4 Holiday Schedule Edit Page

15.3.1 Parameters

The parameters that appear on the Holiday Schedule edit page are as follows

- **Schedule Name:** Displays the name of a new or existing Holiday schedule. You can enter a name in this field for a new schedule or edit it for an existing Holiday schedule.
- **Holidays – Add New Item:** Click Add New Item to add a new holiday to the schedule; the Holiday Name and Date text-entry fields are added to the Holiday Schedule edit page.
- **Holiday Name, Date:** This lets you enter a name and date for a new Holiday schedule. The date format is MM/DD or MM/DD/YY.

  Not entering a value for the year will repeat the same month and day throughout year.

15.3.2 Configuring the Holiday Schedule

To configure the Holiday schedule:

Step 1 Select a holiday from the Schedule Name drop-down list or add a new one by clicking **Add New** and entering a name and date in the Holiday Name and Date text-entry fields.

Step 2 Repeat Step 1 for all known holidays.

Step 3 To delete a holiday from the schedule, click Delete Item.

Step 4 Click **Save** to save the Holiday schedule to the database.

15.4 Custom Schedule

The Custom Schedule edit page is shown in Figure 15-5.

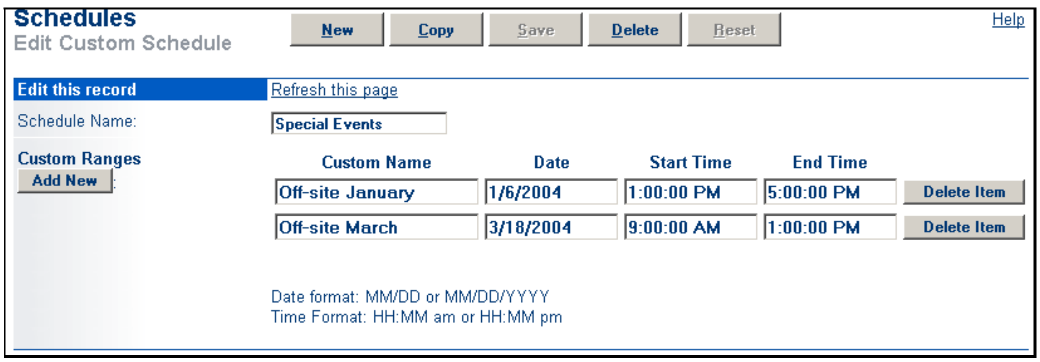


Figure 15-5 Custom Schedule Edit Page

15.4.1 Parameters

The parameters that appear on the Custom Schedule edit page are as follows:

- **Schedule Name:** This displays the name of a new or existing Custom schedule. You can enter a name in this field for a new schedule or edit it for an existing Custom schedule.
- **Custom Ranges – Add New:** Click Add New to add a new Custom schedule; the Custom Name, Date, Start Time, and End Time text-entry fields are added to the Edit Custom Schedule page.
- **Custom Name:** This displays the name of a Custom schedule. Enter the name of a new Custom schedule or edit the name of an existing schedule in this field.
- **Date:** This displays the date when a Custom schedule is used. Enter the date a new Custom schedule or edit the date of an existing schedule in this field. The format is MM/DD or MM/DD/YYYY.

  Not entering a value for the year will repeat the same month and day every year.
- **Start Time:** This displays the start time of a Custom schedule. Enter the start time of a new Custom schedule or edit the start time of an existing schedule in this field. The format is hh:mm:am or hh:mm:pm.

- **End Time:** This displays the end time of a Custom schedule. Enter the end time of a new Custom schedule or edit the end time of an existing schedule in this field. The format is hh:mm:am or hh:mm:pm.

15.4.2 Configuring a Custom Schedule

To configure a Custom schedule:

Step 1 Select a holiday from the Schedule Name drop-down list or add a new one by clicking Add New and entering a name and date for the new holiday in the Custom Name and Date text-entry fields.

Step 2 Enter a start time in the Start Time field.

Step 3 Enter an end time in the End Time field.

Step 4 To delete a custom range, click **Delete Item**.

Step 5 Click **Save** to save the Custom schedule to the database

CHAPTER 16

Configuring Workgroups

This chapter provides information about configuring your ShoreTel system for workgroups, such as Sales or Customer Care, within your organization. The topics discussed include:

16.1 Overview

In a large enterprise, small to medium-sized groups can work as a contact center. The ShoreTel Contact Center Solution provides the ability to queue and distribute calls to these workgroups, supports agent and supervisor functions, and reports on workgroup activity.

16.1.1 Call Routing

The ShoreTel system has many different call routing options to allow enterprises to configure their system and telephony services to match their needs. Callers reach the contact center by calling through a dedicated trunk pointed at the contact center, calling a DID or DNIS number directed to the workgroup, or calling and then navigating to an auto-attendant menu option linked to the workgroup.

Call handling options are combined with the system's schedules to provide even greater flexibility after callers reach the workgroup. Four call handling modes are configured for the workgroup that are scheduled for different times of the day. Each of the four call handling modes starts or stops according to the schedules and can invoke different options and transfer callers to different destinations when all agents are unavailable.

16.1.2 Call Distribution

The ShoreTel system provides flexibility to ShoreTel's Enterprise Contact Center for distributing calls to contact center agents. In addition, the system has options for managing the overflow of inbound calls. Inbound calls to a workgroup are managed on the ShoreTel server and distributed to agents in one of four configurable patterns. When no agent is available, calls can go to a mailbox for the workgroup. accessible by all agents or to a queue where calls can be held until an agent is available.

Distribution of the inbound calls is managed based on agent status. When agents are ready for calls, they log in and begin to receive calls. When they complete their day, they log out and calls are no longer delivered. In addition, the workgroup can optionally be configured so that all agents enter a "wrap-up" mode after every call. In "wrap-up" mode, agents remain logged in but do not receive new calls until the configured wrap-up time passes. This allows agents to complete any required updates to the customer records between calls.

The following is a summary of the call distribution and call overflow options:

Call Distribution Options

- Round Robin
- Top Down
- Longest Idle
- Simultaneous Ring

Call Overflow Options

- Hold calls in the queue until an agent is available
- Transfer calls to another workgroup, extension, or external number
- Transfer calls to the shared workgroup mailbox

16.1.3 Call Queuing

The queue gives the administrator of the workgroup additional flexibility in managing his or her call flow. When all agents are busy, not logged in, or do not answer, callers can be directed to a queue where they are held until an agent is available to take their call. The queue offers up to five steps, each of which can be configured for different caller interactions and to enable caller-selected routing.

The following is a summary of the call queuing options:

Queue Step Configuration Options

- Announce the caller's estimated wait time
- Announce the configured prerecorded prompt
- Provide a menu to offer callers transfer options

Supported Menu Functions

- Customer inputs 0–9, *, #
- Transfers to menus, extensions, or mailboxes
- Repeat prompt or hang up the call

Call Queue Control Options

- Each step can be skipped for call routing flexibility
- Callers are on hold between steps for the configured "step time"
- The last step repeats until the call is delivered

Other Features

- Queue is an option only for on-hours call handling
- Callers hear the main site's on-hold music while waiting

16.1.4 Workgroup Communicators

The ShoreTel Communicator applications for workgroup agents and supervisors provide the information the contact center representatives need in order to be effective, and also give them point-and-click control of their voice communications.

The Communicator applications give agents and supervisors real-time call information, including available Caller ID, call duration, and call state. The call's detailed routing information is displayed so that agents know everyone the current caller spoke with in the enterprise before reaching the contact center. Additionally, the contact center's mailbox is displayed to every agent for accessing and handling the callers who chose to leave a message rather than wait for an agent.

Agents and supervisors have access to the real-time Queue Monitor. This application provides current information on what is happening in the contact center queue. It displays the number of callers, specific information about each, and how long people have been waiting.

The Agent Monitor is the supervisor's tool for managing the workgroup agents. It shows the supervisor the current login status of all the agents, whether they are on a call or not, and allows for changing the agent's status from the supervisor's position to manage contact center coverage.

The following is a summary of the workgroup Communicator features:

Communicator applications

- Display Caller ID, call duration, and call state.
- Display detailed routing information for calls.
- Display and access the shared contact center voice messages.
- Provide point-and-click access to the system's call handling features.
- Log in and log out of the workgroup call flow.

Real-time Queue Monitor

- Display a summary of the number of callers waiting and the longest wait time.
- Show a detailed view of the information about each waiting call.
- Display warnings when the number of calls or longest wait time exceeds the supervisor's thresholds.
- Display or control the call handling mode.

Supervisor's Agent Monitor

- Display the current login status of the agents in the workgroup.
- Show whether agents are on a call and how long they have been talking.
- Control agent's login status from the supervisor's position.

16.1.5 Reporting

As calls are received and handled by the agents of the contact center, records are kept to help supervisors manage the call flow and resources. Each call to the workgroup is logged in terms of how long it spends in the queue, how it ends, which agent handles the call, and how long the call takes.

Three different views of contact center activity are reportable at the ShoreTel server using these easy-to-run, standard reporting tools:

- Queue Summary Report
- Agent Summary Report
- Agent Detail Report

Please refer to Appendix E for more information on the workgroup reports.

16.2 Workgroups Description

Workgroups is a ShoreTel entity that performs Automatic Call Distributor (ACD) functions for inbound calls. Calls are routed to a workgroup through an extension which, in turn, routes them to workgroup agents. Each workgroup is assigned an extension, mailbox, and other parameter settings.

Director allows the creation of a maximum of 256 workgroups, with up to 300 members per workgroup. The Workgroups List page displays all workgroups in the system.

16.2.1 Workgroups List Page

To open the Workgroups list page, click the Workgroups link in the navigation frame, shown in Figure 16-1. From this list page, you can add a new workgroup or link to an existing workgroup.

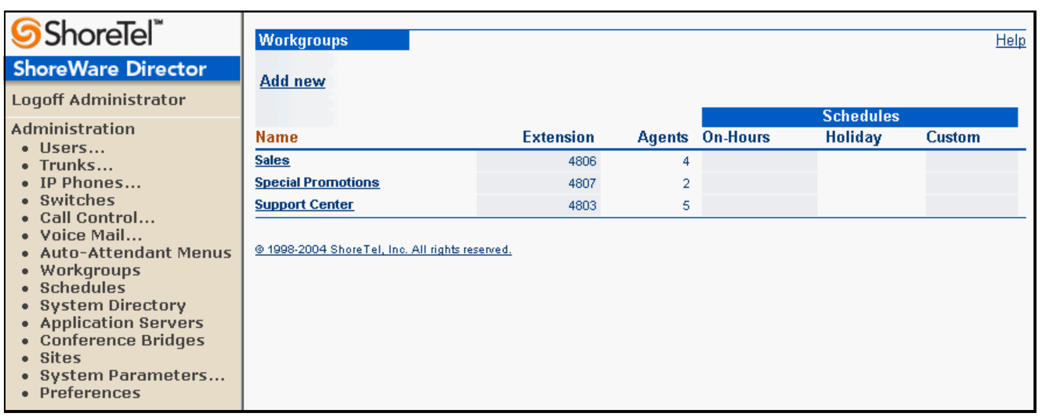


Figure 16-1 Workgroups List Page

The columns on the Workgroups list page are as follows:

- **Name:** This is the name of the workgroup.
- **Extension:** This is the workgroup extension.
- **# Agents:** This value is the number of agents assigned to the workgroup. (Agents can be assigned to multiple workgroups.) The maximum number of agents in a

workgroup depends on whether the Simultaneous Ringing feature has been enabled for the workgroup, as follows:

— 64 agents per workgroup without Simultaneous Ringing

— 16 agents per workgroup when Simultaneous Ringing is enabled

- **On-Hours:** This is the name of the On-Hours schedule, if any, that is associated with a workgroup (named in the Name column at the left).
- **Holiday:** This is the name of the Holiday schedule, if any, that is associated with an existing workgroup.
- **Custom:** This is the name of the Custom schedule, if any, that is associated with an existing workgroup.

16.2.2 Editing Workgroups

This section describes how to edit workgroups in the Workgroups editing page (Figure 16-2). The tasks include assigning extensions and user group features and scheduling.

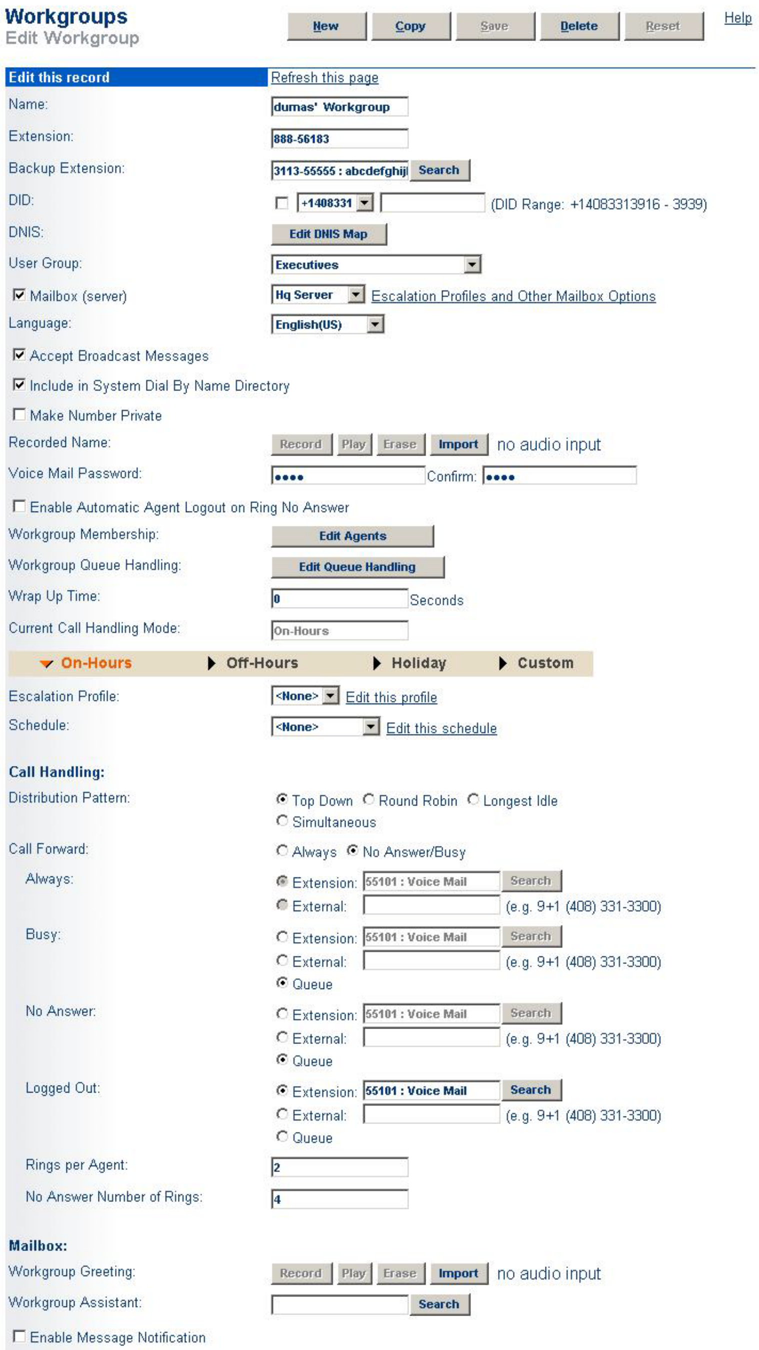


Figure 16-2 Workgroups Edit Page

The parameters on the Workgroups edit page are as follows:

- **Name:** (Required) Each workgroup must have a name.
- **Extension:** (Required) Each workgroup must have a unique extension number. The system can support up to 128 workgroup extensions

  If the system administrator changes the extension number of a user who has an associated mailbox, the user's messages are retained.

This is a required parameter and must be unique. A maximum of 128 workgroup extensions can be defined.

- **Backup Extension:** (Required) The backup extension of the workgroup supports back-up call routing in case of failures. If the workgroup does not answer after a specified number of rings (because of, for example, an unavailable server or network problem), the call is routed to this extension.

  If the call volume is low, calls can be routed to an agent. If the call volume is high, we recommend that these calls go to a "must answer line" with a distinctive ring. Agents can use the call pickup feature to "pick" calls from the "must answer line" if the workgroup server is unavailable.

- **DID:** (Optional) This lets you assign a Direct Inward Dial (DID) number to the workgroup. You can assign one DID number to a workgroup.

  See "Assigning DID Numbers from a Range" on page 165 for additional information about DID numbers and ranges.

- **DNIS:** (Optional) Clicking the ***Edit DNIS Map*** button opens the ***Select DNIS Trunk Group*** dialog box. This box lets you select a trunk group for DNIS routing. DNIS is typically used to route 800-number calls to a workgroup or application. Only trunk groups configured for DNIS appear in the dialog box. You can assign multiple DNIS numbers to a workgroup. See also "Selecting a DNIS Trunk Group for DNIS Routing" on page 490.

- **User Group:** (Required) This lets you assign the workgroup to a user group from the accompanying drop-down list. The workgroup must inherit permission from the User Groups COS, since it has access to some telephony features (such as Call Forward External), as well as some voice mail features (such as Incoming Message Length and Message Notification). See "User Groups" on page 325 and "Telephony Features Permissions" on page 315.

  To use **Barge In**, **Record**, or **Monitor**, users must belong to groups with the necessary permissions enabled in the groups' COS.

- **Mailbox (server):** This provides the workgroup with a mailbox on the associated server. If you change the server, all messages are automatically moved to the new server.

  Calls to the workgroup that go to voice mail end up in the workgroup mailbox. Calls to a user that go to voice mail end up in the user's mailbox. The workgroup mailbox is shared by all workgroup members who run the Workgroup Agent or Supervisor Communicator. In addition, you can log into the workgroup mailbox over the telephone using the mailbox number and voice mail password.

  This is an optional parameter.

- **Language:** Select a language from the drop-down list. This is the language that will be used by the auto attendant for prompts played to calls in the queue, such as "your estimated wait time is..."

- **Accept Broadcast Messages:** This lets the workgroup receive broadcast messages.

  In general, you will want to remove the workgroup from the broadcast message list.

  This is an optional parameter.

- **Include in System Dial By Name Directory**: This includes the workgroup in the auto-attendant's dial-by-name directory.

  This is an optional parameter.

- **Make Number Private:** Checking this check box removes this number from the system directory and call handling destination lists.
- **Recorded Name:** The Record, Play, Enter, and Import buttons let you record a name for the workgroup. This name is used as part of the default mailbox greeting as well as in the Dial By Name directory.

  You can use your PC microphone and speakers or a telephone to play and record within ShoreTel Director. Please refer to the Auto-Attendant options for more information.

  You can also import prompts into ShoreTel Director. Prompts must be recorded as µ-law, WAV files.

  This is an optional parameter.
- **Voice Mail Password:** This lets you enter and confirm the workgroup's voice mail password.

  The voice mail password is used in conjunction with the mailbox number to log in to the workgroup mailbox over the telephone.

  This is a required parameter. The initial default is "1234". Passwords may be numeric only.
- **Enable Automatic Agent Logout On Ring No Answer:** This check box lets you automatically log out agents that do not answer a workgroup call after a specified number of rings.

  This feature is useful to avoid calls repeatedly being offered to an agent who has physically left the workgroup but has forgotten to log out.

  This is an optional parameter.
- **Workgroup Membership:** (Required) The Edit Agents button invokes the Workgroup Membership page, which lets you add and remove agents to and from the workgroup. You can also change the call distribution hunt order of agents via this page.
- **Workgroup Queue Handling:** (Required) The Edit Queue Handling button invokes the Workgroup Queue Handling edit page. This page lets you edit the Queue Step Menu as well as the Queue Thresholds. For more information, see "Editing Workgroup Queue Handling" on page 492.
- **Wrap Up Time: Wrap Up Time** allows agents a fixed period of time in seconds to complete post-call tasks before being presented with another call. When the call wrap-up time is set to zero, this feature is effectively disabled.

  This is an optional parameter.
- **Current Call Handling Mode:** The is a read-only display of the current call handling mode as defined by any associated schedules.
- **Escalation Profile:** Select an Escalation Profile from the drop-down menu to associate the desired profile with this workgroup. (See "Configuring Escalation Notification" on page 452 for details.)
- **Schedule:** You can configure schedules against the On-Hours, Holiday, and Custom modes that automatically change the call handling of the workgroup. The rules for schedules are:

  — If it is custom time, use **Custom** mode;

  — If it is holiday time, use Holiday mode;

— If it is on-hours time, use On-Hours mode;

— Otherwise, use Off-Hours mode.

If no schedules are specified, On-Hours mode is used.

The Edit **this** schedule link provides a quick way to navigate to the associated schedule. For tips on editing schedules, refer to Chapter 15: Configuring Schedules starting on page 475.

This is an optional parameter.

- **Call Handling Distribution Pattern:** (Required) This lets you configure how calls are distributed to agents within the workgroup:

  — **Top Down** begins at the top of a list of agents and proceeds down the list looking for an available agent until one is found.

  — **Round Robin** selects the next agent on the list following the agent that received the last call and then starting again at the top of the list and so on. This works in a looping fashion.

  — **Longest Idle** distributes calls to the agent who has been idle the longest.

  — **Simultaneous** distributes calls simultaneously to all available agents, so all phones for the workgroup ring simultaneously.

- **Call Forward:** These buttons let you specify when calls are forwarded. The conditions are ***Always*** or ***No Answer/Busy***. The default is ***No Answer/Busy***.

  The Always condition forwards calls to the number specified in the Always Destination parameter immediately when a call is received.

  The No Answer/Busy condition forwards calls:

  — to the No Answer Destination after the specified number of rings, or

  — to the Busy Destination immediately if the user's call stack is full.

  This is a required parameter.

- **Always:** When the Always call forward option is selected, calls are forwarded immediately to this Extension. You can also forward calls to an External number (access code required). If the Call Forward parameter is set to Always, enter extensions here.

- **Busy:** When the Busy call forward option is selected, calls are forwarded to this extension immediately if all agents are busy. You can also forward calls to Queue, an External number (access code required), or Extensions here.

- **No Answer:** When the No Answer call forward option is selected, calls are forwarded to this extension after the specified number of rings. You can also forward calls to Queue, an External number (access code required), or Extensions here.

- **Logged Out:** When the Logged Out call forward option is selected, calls are forwarded to this extension if all agents are logged out of the workgroup. You can also forward calls to Queue or to an External number (access code required).

- **Rings per Agent:** This is the number of rings attempted before the call is forwarded to the next available agent.

- **No Answer Number of Rings:** This is the maximum number of rings that a call will ring before it is forwarded to the no-answer destination.

- **Mailbox / Workgroup Greeting:** The Record, Play, Enter, and Import buttons let you record a mailbox greeting for each call handling mode of the workgroup.

  This is an optional parameter.

- **Workgroup Assistant:** To assign a workgroup assistant, select one from the drop-down list. When a caller is connected to the workgroup's voice mail and enters "0", the call is transferred to the workgroup assistant extension.

  This is an optional parameter.

- **Enable Calling Message Notification:** This check box activates the message notification feature for the workgroup's mailbox.

  This is an optional parameter.

16.3 Selecting a DNIS Trunk Group for DNIS Routing

To add Dialed Number Information Services (DNIS) routing to a trunk group:

Step 1 Click **Edit DNIS Map** in the Workgroups edit page. The DNIS Trunk Group list opens (Figure 16-3).

Step 2 Click an available trunk group from the list, and then click **OK**.

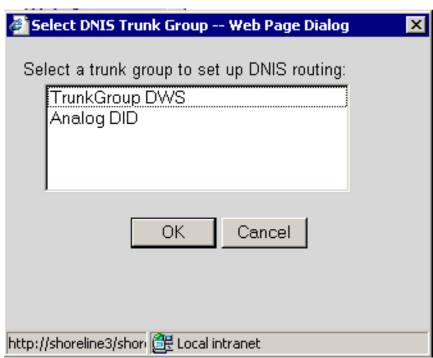


Figure 16-3 Select DNIS Trunk Group Dialog Box

16.4 Editing Workgroup Membership

To build a list of workgroup members, click Edit Agents found on the Workgroups edit page. The Workgroup Membership edit page (Figure 16-4) lets you add and remove agents to and from the workgroup.

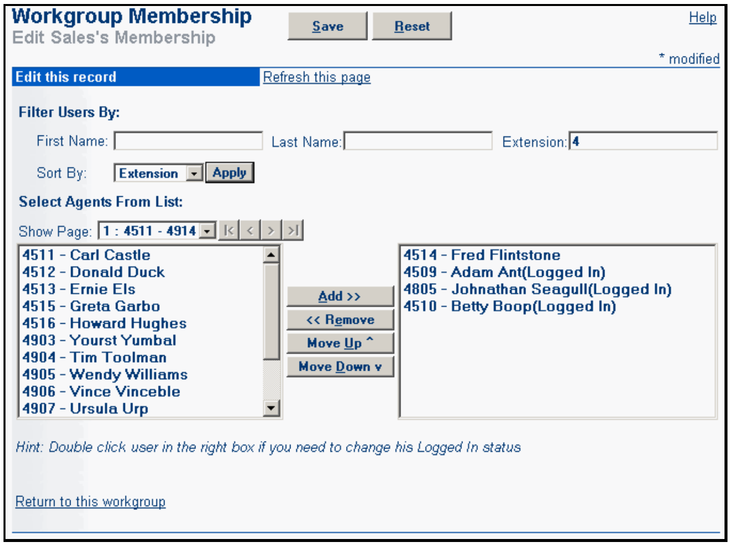


Figure 16-4 Workgroup Membership Edit Page

This page lets you add and remove agents to and from the workgroup, and also change the order of agents. Note that if more members exist than can be displayed on one page, you can use the forward and back buttons to scroll through the members or enter filter criteria in the Filter Users By box. Also note that you can sort members by Extension, First Name, or Last Name by using the Sort By drop-down list.

To add members to a workgroup, select a member's name from list on the left side of the page and click Add. The member's name appears in the list on the right. When you add a new workgroup member, that member is in a logged out state by default.

To remove a member from the workgroup, select that member's name from the list at the right side of the page and click Remove. The member's name is returned to the list on the left side of the page.

To change the order of a member's hunt order in the workgroup, select the member's name in the **right-most** list and click Move Up or Move Down until the member appears in the order you want.

To change the member's login status, double-click the name from the **right-most** list and the Edit Workgroup Member dialog box (Figure 16-5) appears.

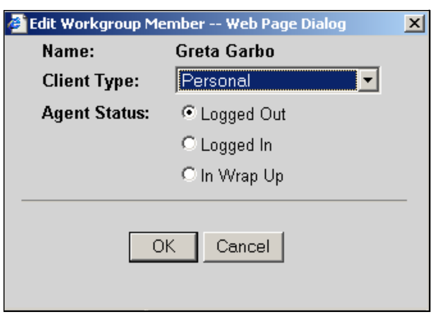


Figure 16-5 Edit Workgroup Member Dialog Box

You can change the agent's Client Type, Logged In, or Call Wrap Up status, and click OK to commit the changes. The change is reflected in the Name list on the Workgroup Membership edit page.

16.5 Editing Workgroup Queue Handling

The Workgroup Queue Handling edit page (Figure 16-6) lets you edit the queue threshold and the Queue Step Menu. Click Edit Queue Handling from the Workgroups edit page.

A calling party that reaches the queue will hear the following, in order:

1. The recorded prompt.
2. Silence for eight seconds during the "wait for digits" time-out.
3. The estimated wait time (if enabled).
4. Music on hold for the configured duration.

The calling party does not have access to the DTMF actions while listening to music on hold.

The ShoreTel system ignores DTMF actions if there is no message for the queue step.

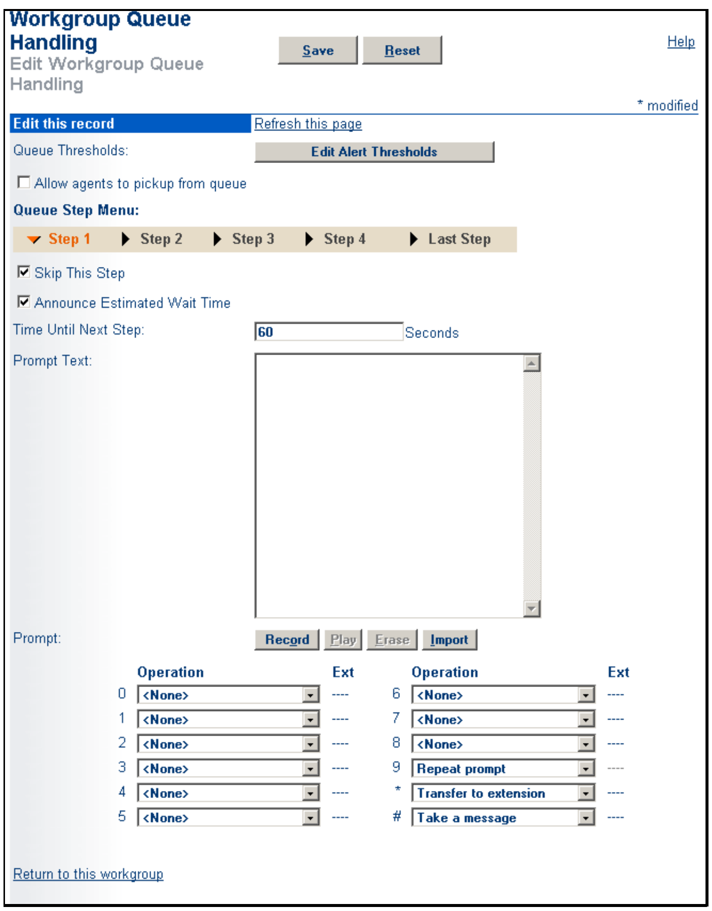


Figure 16-6 Workgroup Queue Handling Edit Page

The "Allow agents to pickup from queue" checkbox enables and disables the queue handling step menu.

A five-step menus can be set for each queue.

To set the Queue Step Menu:

Step 1 To set a step, click Step 1, Step 2, Step 3, Step 4, or Last Step. To skip a step, select Skip This Step.

Step 2 Select **Announce Estimated Wait Time** to enable this feature. This informs the caller of how much time is left in queue.

Estimated wait time is a moving average based on the duration of the previous calls and rounded to the minute. It is calculated as:
*Average wait seconds = ((“Average wait seconds” * 9) + “New wait time”) / 10*

Where the “New wait time” is the time it took the last call to get to an agent. The wait time that is spoken to the caller is:
*Spoken wait time = “Position in queue” * “Average wait seconds”*

This means the wait time is based on a rolling weighted average of previous calls. After 10 calls, 61% of the time is based on the 10 most recent calls. After 20 calls, 86% of the time is based on the last 20 calls. The wait time may be inaccurate with a low call volume.

Step 3 Enter the value for the Time Until Next Step routine in the Seconds field.

Step 4 Enter the text used for the prompt in the Prompt Text field.

Step 5 Click **Record** to record the prompt. If you have a “canned” prompt, click **Import** to import the Wave file.

Click **Play** to hear the recording. Click **Erase** to erase the recording.

Prompts on the ShoreTel system can be imported into the system using µ-law, Wave file format. If you would like your prompts to match the voice of the ShoreTel system, please contact Worldly Voices at www.worldlyvoices.com and request that “Connie” record your prompts. Worldly Voices provides this service with a rapid turnaround time for a nominal fee.

Step 6 Click an operation function from the **Operation** drop-down list and assign it an extension. The Single Digit Actions allow you to configure all the digits of the telephone keypad (0–9, #, *) for operations similar to an auto-attendant menu. The following actions are supported:

- None
- Repeat prompt
- Go to menu
- Transfer to extension
- Take a message
- Hang up

Details:

- For information about overflowing / interflowing calls to a workgroup call, see "Configuring Workgroup Overflow / Interflow" on page 494.

16.5.1 Configuring Workgroup Overflow / Interflow

Workgroup overflow and interflow capabilities offer a way to reduce the wait time for callers who are dialing into an Automatic Call Distribution (ACD). This helps to ensure faster service and greater customer satisfaction.

Calls can be overflowed (i.e. transferred from one workgroup queue to another higher priority queue once a wait time threshold has been exceeded) in order to ensure that certain customers who have paid for a higher level of support service (e.g. “Gold Level”) have their calls answered faster.

Alternatively, calls can be interflowed to any dialable number when a wait time threshold has been exceeded. This external number could be, for example, a cell phone number for a supervisor who could answer the call immediately.

Interflowing a call is typically done as the final step in a series of overflows. In other words, if a call is sent from one workgroup queue to another without being answered, the call is interflowed from the Last Step to an external number (such as the supervisor's cell phone).

To configure the Workgroup Overflow / Interflow feature:

Step 1 The overflow / interflow functionality is set as the Last Step within the Queue Step Menu. Thus, you must follow the procedure in "Editing Workgroup Queue Handling" on page 492 before configuring the overflow / interflow option.

Step 2 Click on the **Last Step** tab and scroll down to the Overflow / Interflow section toward the bottom of the window, as shown in Figure 16-7.

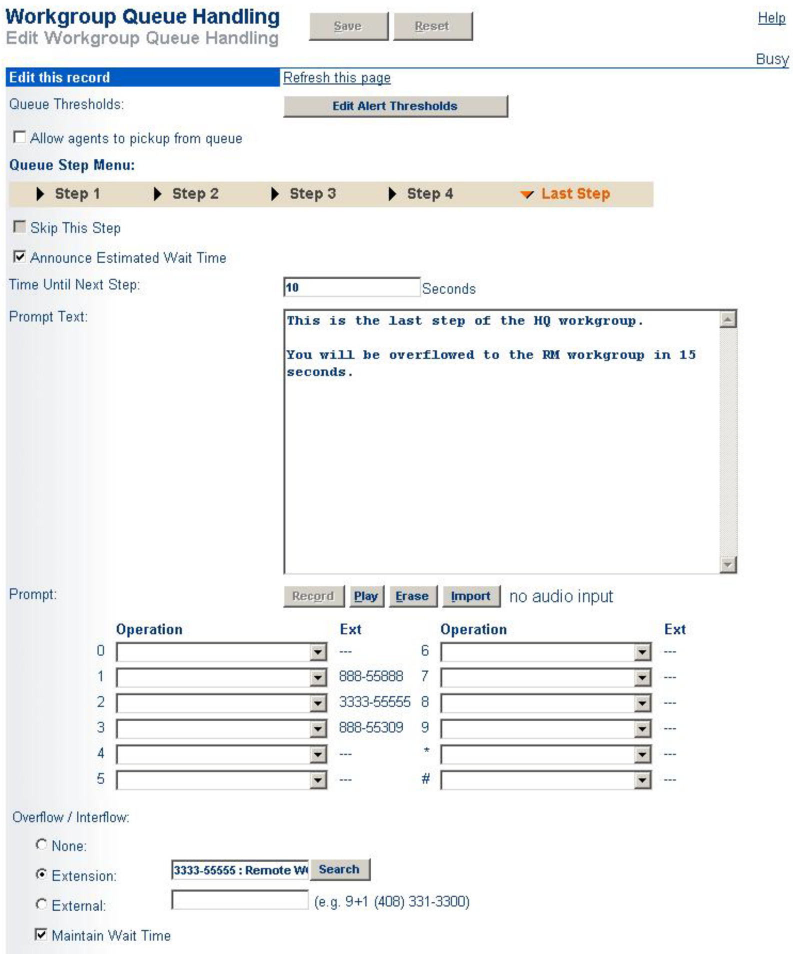


Figure 16-7 Overflow / Interflow options found at Last Step of WG Queue Handling

Step 3 In the **Overflow / Interflow** section, select the desired radio button:

- Selecting **None** results in the Overflow / Interflow behavior being disabled.
- Selecting **Extension** enables the Overflow or Interflow behavior. If this is selected, click the Search button to locate the desired extension where calls will be overflowed or interflowed.
- Selecting **External** enables the Interflow behavior. Enter the desired dialable number where calls are sent when the threshold is exceeded.

Step 4 Select the **Maintain Wait Time** check box to store information about the length of time a caller has been waiting. The starting time associated with a call will be preserved even after the call has been transferred from one workgroup queue to another, and the caller's "place in line" will be reserved within the next queue. If the check box is disabled, the wait time will be lost when the caller is transferred and the caller will go to the back of the line in the new queue.

If the call is interflowed to an external number, the wait time cannot be retained. The system will recognize if these conditions are present and the **Maintain Wait Time** check box will be grayed out.

Step 5 Click **Save** to store your changes.

Details: The number of seconds in the Time Until Next Step field for Steps 1 through the Last Step, when added up, represent the total amount of time the caller waits before being overflowed / interflowed.

16.6 Workgroup Thresholds

To adjust the queue threshold, click Edit Alert Thresholds found on the Workgroup Queue Handling page. The Workgroup Thresholds page (Figure 16-8) appears.

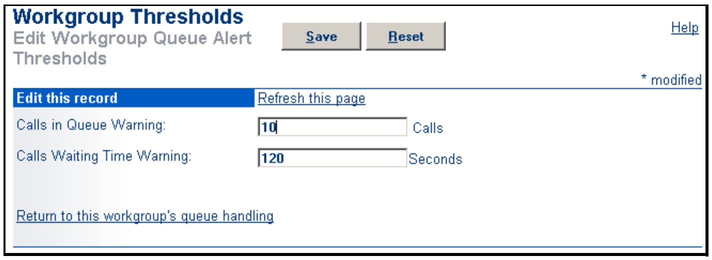


Figure 16-8 Workgroup Thresholds Page

The threshold of **Calls in Queue Warning** specifies the number of calls in queue that triggers an alert in the Queue Monitor in the workgroup Communicator application.

The Calls Waiting Time Warning specifies the number of seconds that triggers an alert in the Queue Monitor for the longest wait time of any caller in the workgroup queue.

16.7 Multi-site Workgroups

Multi-site Workgroups is a feature that allows organizations to deploy Workgroups functionality at multiple sites ensuring that enterprises don't lose any key applications needed to drive revenue and customer support. Multi-site Workgroups enables a group of users to appear as a single unit to calling parties, by distributing calls to Workgroup agents at multiple sites without relying solely on the Headquarters server. When successfully deployed, Multi-site Workgroups will provide the reliability and availability required for organizations to remain competitive.

16.7.1 Configuring Multi-site Workgroups

16.7.1.1 Site Specific Workgroups

Using Multi-site Workgroups allows more flexibility through the use of backup Workgroups where the Workgroups can be hosted on a different server. When a call is directed to a Workgroup the switch will route the call to the Workgroup Extension. If the Workgroup extension is unreachable because of server or WAN failure, the switch will route the call to the Backup Extension in the server that was configured to handle that workgroup. The Backup Extension can be a Workgroup, Hunt Group, Menu or any system extension.

NOTE This routing process may take a brief period of time until the no answer number of rings limit is reached which will then trigger the activation of the backup extension.

To configure a site specific Workgroup, follow these steps.

Step 1 Log into ShoreTel Director and select Workgroups.

Step 2 Select **New** to create a new Workgroup or select an existing Workgroup that you wish to modify. See Figure 16-9.

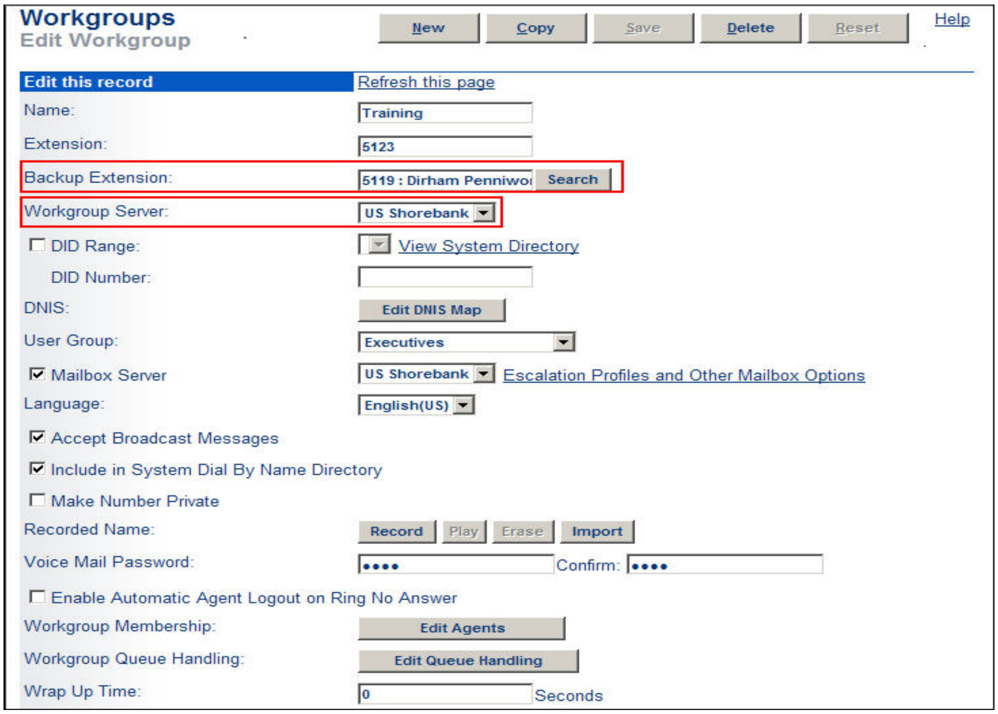


Figure 16-9 Workgroups Edit Screen

Step 3 Edit the following fields in the Edit Workgroup screen.

- **Workgroup Server** - The server hosting the Workgroup. All Distributed Voicemail Servers will be listed. Voicemail switches will not be listed.
- **Backup extension** -The backup extension of the Workgroup. If the Workgroup does not answer after the specified number of rings (for example, server unavailable or network problem), the call is routed to this extension. This allows you to configure backup call routing in case of failures.

  NOTE If the call volume is low, you can route these calls to an agent. If the call volume is high, it is recommended that you route these calls to a must answer line with a distinctive-sounding ring. Agents can use the call pickup feature to pick calls from the must answer line if the Workgroup server is unavailable.

16.7.2 System-Wide Workgroups

System Wide Workgroup is the setup in which two or more servers are providing call-routing access. A System Wide Workgroup is supported by Hunt Groups along with Multi-site Workgroups. Together, they offer calls first to the primary Workgroup and then to other Workgroups that are members of the Hunt Group.

Hunt Groups can have up to 16 members, which provide a higher degree of resiliency. Using Hunt Groups is a faster method for call routing in case of a failure because you can configure Hunt Groups to offer calls to a larger number of backup Workgroups.

To configure a System Wide Workgroup, follow these steps.

Step 1 Log into ShoreTel Director and select Call Control -> Hunt Groups.

Step 2 Click **Administration > Call Control > Hunt Groups**. The Hunt Groups page appears.

Step 3 Select **New** to create a new Hunt Group or select an existing Hunt Group that you wish to modify. The Edit Hunt Groups page appears as shown in Figure 16-10.

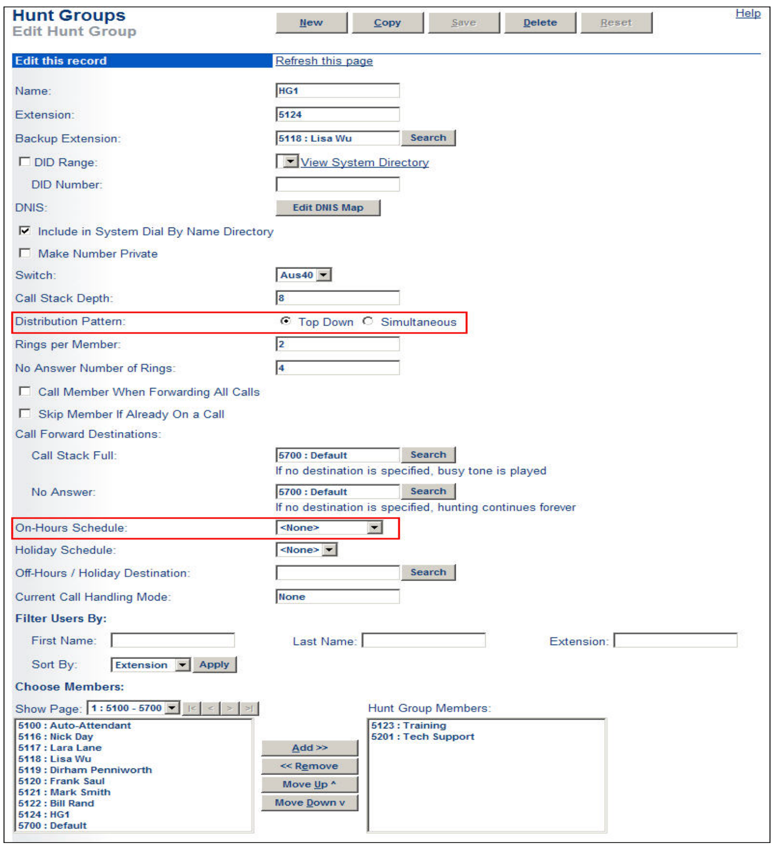


Figure 16-10 Hunt Groups Edit Screen

Step 4 Edit the following fields in the Edit Hunt Group screen.

- **Distribution Pattern** - Click either the Top Down or the Simultaneous radio button. With Top Down, the system sequentially hunts through the ordered list of group members. With Simultaneous, the phones for all group members ring simultaneously. With Simultaneous, the first group member to answer the phone gets the call. The default is Top Down.

  NOTE: Simultaneous and Top Down are mutually exclusive.

- **On-Hours Schedule** -From the drop-down list, select an on-hours schedule or None. Selecting None causes all calls to be treated as if it is on-hours.

16.7.3 Important Considerations

16.7.3.1 Switches that Provide Voice Mail

This feature does not work on the voice switches that provide voice mail (ShoreTel Voice Switch 90V, ShoreTel Voice Switch BRIV, and so on).

16.7.3.2 CDR

All CDR records collected by remote severs will be sent to the Headquarters server. The existing TMSCDR DCOM interface will be used by the remote servers to send records to the Headquarters server. If the Headquarters server is offline then records will be queued on remote servers until the Headquarters server comes back on line. There will be a time limit on the CDR records that will be kept by remote servers when the Headquarters server is offline. The Workgroup server will use the same time limit that is used by TMS.

16.7.3.3 Call Handling Mode Scheduling

Workgroups can have their call handling mode changed based on a system schedule. For Multi-site Workgroups, call handling scheduling is based on the time zone of the server hosting the Workgroups. Local call handling mode changes can not be made by any Communicator application or IP phone.

16.7.3.4 Hunt Group and Workgroup Scheduling

An additional consideration is the scheduling of the activation time for both Hunt Groups and Workgroups.Hunt Group schedules times are based on the Headquarters server time. Workgroups schedule times are based on the Workgroups server's time. Scheduling for Workgroups and Hunt Groups are configured by calculating the time zone offset necessary for the Hunt Group to activate at the same time as the Workgroup. Refer to the ShoreTel Administration Guide for more information on Hunt Group and Workgroup scheduling.

16.7.3.5 Headquarters Server Offline

If the HQ server is down, then changes to Agent state do not work because the HQ server is hosting all of the agents' states. All states, whether logged in, logged out, or in wrap up, remain unchanged until the HQ server is back on-line.

CHAPTER 17

Managing the System Directory

This chapter discusses how the system directory is used and provides instructions for adding new entries and editing existing ones. The topics discussed include:

- "System Directory" on page 501
- "Adding an Off-system Directory Entry" on page 503

17.1 System Directory

The system directory is a company-wide address book that list users and off-system contacts by name and provides additional information such as home address and phone numbers. This directory is read-only to general users through the ShoreTel clients. Only the system administrator can update it using ShoreTel Director. Users can copy system directory entries to their personal directories.

Individual users automatically populate the system directory when you create them. (For more information about creating system users, see the "Individual Users" on page 329). You may also add additional listing to the system directory using the System Directory page in ShoreTel Director.

To access the System Directory page:

Step 1 Launch ShoreTel Director with administrator privilege.

Step 2 Click **Administration > System Directory**. The System Directory page appears as shown in Figure 17-1.

Figure 17-1 System Directory Page

ShoreTel Communicator automatically populates each user's Quick Dialer with entries from the system directory, the user's personal directory, and all Microsoft Outlook Contact folders. This includes each user's personal contacts as well as any contacts on the Microsoft Exchange Server.

To access the system directory, click System Directory from the navigation frame.

The system directory can be filtered by site or dial number. You can also page through the directory by page, with a configurable number of records per page displayed.

The columns that appear in the table on the System Directory page are as follows:

- **First Name**: This is the first name of an existing directory entry.
- **Last Name**: This is the last name of an existing directory entry.
- **Ext.**: This is typically a contact's telephone number or extension.
- **Type**: This is the type of extension, such as workgroup, FAX, and so on.
- **Site**: This is the site where the extension is located.
- **Trunk Group**: This is the trunk group associated with the extension.
- **DID**: This is the direct inward dial number for the user.
- **Work**: This is the work number for the user.
- **Home**: This is the home number for the user.

- **Fax:** This is the FAX number for the user.
- **Cell:** This is the cell number for the user.

17.2 Adding an Off-system Directory Entry

You can add commonly accessed off-system contacts to the directory using the System Directory page.To add new off-system contact, do the following:

Step 1 Launch ShoreTel Director with administrator privilege.

Step 2 Click **Administration > System Directory**. The System Directory page appears.

Step 3 In the By Site field, select the site where you want the listing to appear.

Step 4 Click the **New** button. The Edit Entry page appears shown in Figure 17-2.

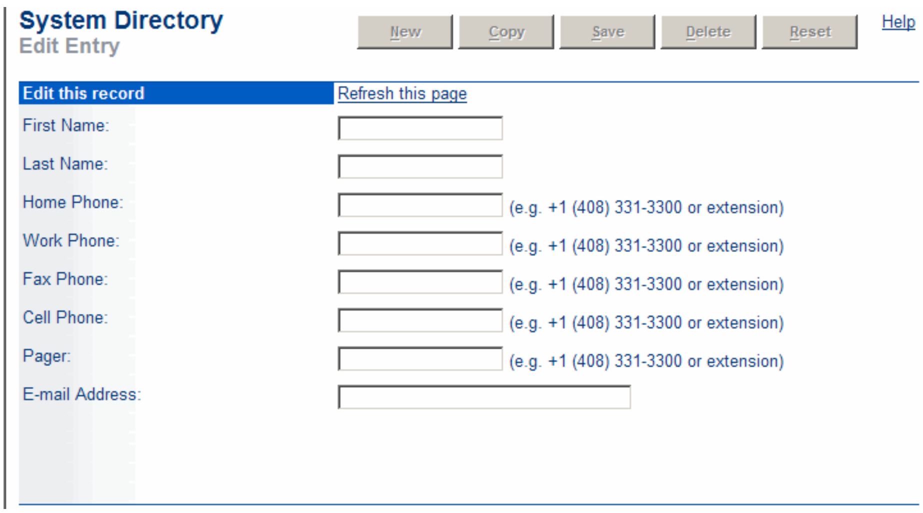


Figure 17-2 System Directory Edit Entry Page

Step 5 Enter the pertinent information in the following fields as required:

- **First Name:** This field displays the user's first name.
- **Last Name:** This field displays the user's last name.
- **Home Phone:** This is the user's home telephone number. Use the format shown next to the field to make this entry.
- **Work Phone:** This is a work telephone number for the user other than his or her extension. Use the format shown next to the field to make this entry. Do not enter a user extension number in this field.
- **Fax Phone:** This is the user's FAX number. Use the format shown next to the field to make this entry.
- **Cell Phone:** This is the user's cellular telephone number. Use the format shown next to the field to make this entry.

- **Pager:** This is the user's pager number. Use the format shown next to the field to make this entry.
- **E-mail Address:** This is the user's e-mail address.

Step 6 Click **Save**.

17.2.1 Go to This User Link

The "Go to this user" link appears on the System Directory edit entry page when you are editing an existing user directory entry. This link takes you to the Edit User page so that you can edit the user's general, personal, and distribution list options.

CHAPTER 18

Session Initiation Protocol

This chapter provides detailed information about the Session Initiation Protocol (SIP). You should refer to this chapter for help in planning a SIP deployment on your ShoreTel system. The topics discussed include:

- "Setting Up SIP Trunks" on page 505
- "Setting Up SIP Extensions" on page 518
- "Setting Up SIP Integration with Third-Party Unified Messaging Systems" on page 532

ShoreTel supports SIP trunks and SIP Extensions

18.1 Setting Up SIP Trunks

Session Initiation Protocol (SIP - RFC 3261) is a newer protocol that is still being fine tuned by the IETF and that is regarded as having the potential to become the global signaling standard that will enable all switches, gateways, and phones to talk to one another.

The protocol, which works at the application layer, allows users to initiate interactive sessions between any network devices that support the protocol. SIP is capable of initiating or terminating Internet telephony calls and other multimedia applications such as video or gaming.

The protocol is based on a client-server model. With support for redirection services, networked users can initiate a call or receive a call, regardless of their physical location.

In its networking negotiations SIP takes into account the following pieces of information:

- The address of the end system.
- The physical media.
- The call recipient's acceptance to the invitation.

The protocol then configures the parameters for the session and handles the call setup and tear-down.

SIP allows two discrete ShoreTel systems to be integrated with any IP connection, without the need for physical tie trunking. (Note that care should be taken to make sure that the extension numbering plans in the two systems do not overlap, and that if they do overlap, translation tables need to be used to resolve conflicts.)

Further, the addition of SIP obviates the need to support other trunking standards, such as BRI, through use of a SIP gateway.

SIP trunks will be assigned to a particular switch as with any other trunk, so that SIP calls into and out of the ShoreTel system will be routed through these trunks. However, up to five SIP trunks can be associated with one analog switch port, meaning that there will be no physical channel/port associated with each SIP trunk. The SIP trunk is a logical trunk end point which only handles call control responsibilities. The media flows directly between the end-point SIP devices (i.e. call initiator and the call terminator), freeing the switch from the burden of controlling media flow.

18.1.1 Supported RFCs

ShoreTel supports SIP Trunks as specified by the following RFCs:

- 1889 - Transport Protocol for RTP Applications
- 2806 - URL for Telephone Calls
- 2327 – Session Description Protocol (SDP)
- 2396 - URI (Uniform Resource Identifiers)
- 2833 - DTMF
- 2976 - SIP Info
- 3261 - SIP (Session Initiation Protocol)
- 3361 - DHCP (for IPv4)
- 3515 - SIP Refer Method
- 3891 - SIP Replaces Header
- 3892 - SIP Referred-by Mechanism
- 3966 - URI for Telephone Calls

18.1.2 General SIP Comments

Previous ShoreTel versions support SIP trunks in the following configurations:

- Static Trunking connects two PBXs within a ShoreTel installation.
- Dynamic Trunking connects a PBX to a SIP device.
- Outbound SIP Trunking connects a PBX to an Ingate SIP Trunk Module, which can connect to external IP networks or to an ITSP to for connecting to the PSTN.

The SIP trunking implementation in previous versions does not support the use of SIP devices as ShoreTel extensions.

18.1.2.1 Conferencing

- Ports for MakeMe conferences must be available on the initiating side of a 3-way conference call involving a SIP end-point.
- MakeMe conference ports are needed even for 3-way conference. Note that configuration of any MakeMe conferencing support in Director requires a minimum of 3 available conference ports.
- An individual SIP trunk must be provisioned for each call to the SIP device (including conference-in or transferred calls). Thus, static SIP trunks must be provisioned with additional trunks in line with the highest anticipated number of such calls. Similarly, dynamic SIP trunks also require that additional individual dynamic SIP trunks are provisioned to handle calls that are placed on hold or for conference-in calls.

18.1.2.2 DTMF

- ShoreTel can be configured to use SIP INFO for DTMF signaling in environments where out-of-band DTMF is needed but RFC 2833 is not applicable. SIP tie trunks must use SIP INFO and cannot use RFC 2833 DTMF Relay.
- ShoreTel supports RFC2833 (DTMF) for users calling over SIP trunks regardless of the negotiated voice codec.

18.1.2.3 Foreign Language Support

- In addition to English, ShoreTel will support Spanish, French and German (Caller Name, Called Name, User Name) over SIP tie trunks and service provider trunks, although certain third-party devices may not be able to display the Spanish or German characters.

18.1.2.4 Routing with Static and Dynamic Trunks

From the trunk group perspective, when static and dynamic trunks are used:

- A switch can have only one trunk group with dynamic trunks is.
  - — Outbound calls to this trunk group must be completed based on the registration table.
  - — Calls to the same IP address do not work.
  - — Calls to different devices going to the PSTN are randomly selected.
- Trunk groups with static IP addresses will not route calls based on OSE ranges due to the fact that static trunks do not need registration.
  - — The switch sends the call to the next available trunk instead of sending it to the correct OSE within the range.
  - — This issue can be solved by creating a trunk group on a per-device basis.
- OSEs routed over trunk groups in more than one switch (with dynamic trunks) fail.

18.1.2.5 Extension Assignment over SIP Trunks

ShoreTel supports Extension Assignment phone assignment to devices available over SIP trunks. Extension Assignment procedures and capabilities over SIP trunks are identical to those for reassignments to phones over other supported trunks.

Using Extension Assignment over SIP Trunks requires using an ITSP that supports SIP-INFO. If the ITSP does not support SIP-INFO, ShoreTel cannot support DTMF for users configured for the SIP Trunk who during Extension Assignment calls. This requires that they are configured for “Accept on Answer” and restricts them from using keypad features during the call. Theses users can still control their calls through Communicator.

The following limitations apply:

- SIP Tie lines to other ShoreTel systems do not support Extension Assignment.
- SIP Trunks to supported SIP carriers through Ingate support Extension Assignment.
- SIP Trunks using SIP gateways supported by ShoreTel (including SIP-BRI 8 and Vegastream SIP gateways) support Extension Assignment.

- Gateways must support, and be configured to send, DTMF in INFO messages to support certain functions through Extension Assignment, including Wait for DTMF, ** flash, and ## hangup).
- Extension Assignment users cannot hear in-band call-waiting tone for a second call over SIP Trunks. Communicator can signal the presence of a second call.
- Features not supported by normal calls on a SIP trunk are also not supported by Extension Assignment extension using a SIP trunk.

18.1.2.6 General Feature Consideration

- ShoreTel supports Music On Hold (MOH) over SIP trunks. The capacity limits of MOH switches is the same as other trunks; a switch can provide up to 15 streams. However, these streams can be to other switches or to SIP devices.
- If the ShoreTel server has a conference bridge 4.2 installed, you should not enable SIP. The conference bridge is not compatible with a ShoreTel system that has SIP enabled due to the dynamic RTP port required for SIP.
- Three-way conference on a SIP trunk call uses Make Me conference ports. A minimum of three Make Me ports must be configured to support three-way conferencing.
- A SIP trunk can be a member of a 3-party conference but cannot initiate a 3-way conference (unless the SIP device merges the media streams itself).
- ShoreTel SIP supports basic transfers (i.e. blind transfers) and attended transfers (i.e. consultative transfers).
- Silent Monitoring and Silent Coach are not supported on a SIP trunk call.
- Barge-In is not supported on a SIP trunk call.
- Call recording is not supported on a SIP trunk call. Call recording requires presence of a physical trunk in the call.
- Call redirection by SIP devices is not supported.
- Silence detection on trunk-to-trunk transfers is not supported since it requires a physical trunk.
- Fax (and modem) redirection is not supported with SIP trunks as only physical trunks can detect fax tones.
- SIP Unified Messaging (UM) can support up to 37 concurrent SIP UM calls from IP phones.

18.1.2.7 Additional Configuration Considerations

- SIP Info configuration in a Trunk Group should be enabled if ShoreTel SIP tie trunks are used.
- Overlapping number plans are not allowed between two systems tied with SIP trunks unless digit translation is used.
- When translating digits between two ShoreTel systems tied with SIP trunks, even system extensions like VM, AA should be properly translated.
- SIP devices should either be physically present in the ShoreTel site where the ShoreTel switch is hosting the SIP trunk or should be out of the ShoreTel network.

- A SIP trunk group cannot host both dynamic and static SIP trunks simultaneously.
- A SIP trunk group hosting dynamic SIP trunks cannot span ShoreTel switches.
- When ShoreTel is working with Dynamic Trunks:
  - — Multiple registrations of different numbers using the same IP address is not supported as ShoreTel uses the last one received. (This is the case of Mediatrix 2102/1402; customers are expected to use only static trunks for these devices.)
- When ShoreTel is working with Static Trunks:
  - — OSE ranges might not work when different SIP devices are part of the same trunk group. Customers are expected to create a dedicated trunk group for each device that needs a static trunk.
  - — Customer must ensure that SIP devices work with static trunks. Routing problems may occur when the same switch has a dynamic trunk group. These devices should not be registered with the ShoreTel system.
- Director does not show information about the SIP devices registered in a switch. This information can be accessed by telneting to the switch and issuing the command print_register_table (applies only to dynamic SIP trunks).
- Groups of SIP trunks can be created at once but are deleted one at a time.
- SIP devices need to work with dynamic audio ports. Customers are expected to disable the parameter that forces the system to use only audio port 5004.
  - — This parameter is found in Director under Call Control > Options.

18.1.3 Configuring SIP Trunks

Configuring SIP on your ShoreTel system consists of the following tasks:

- Configuring the ShoreTel System via Director
  - — Reserve the Trunk.
  - — Create a Trunk Group.
  - — Create a Trunk (static or dynamic).
- Configuring the SIP Device (per the manufacturer's instructions).

These tasks are discussed in more detail in the sections that follow.

18.1.3.1 Reserving SIP Trunk on ShoreTel Switches

One of the first tasks involved in configuring your ShoreTel system for Session Initiation Protocol (SIP) is to reserve the trunk. To reserve a new SIP trunk, do the following:

Step 1 Launch **ShoreTel Director**.

Step 2 Click **Administration > Platform Hardware > Voice Switches/Service Appliances > Primary**. The Primary Switches page appears.

Step 3 Select the switch on which you want to reserve SIP trunks or add a new switch. (See Chapter 5, "Configuring Switches", starting on page 89 for information about adding switches.) The Switches page as shown in Figure 18-1 appears.

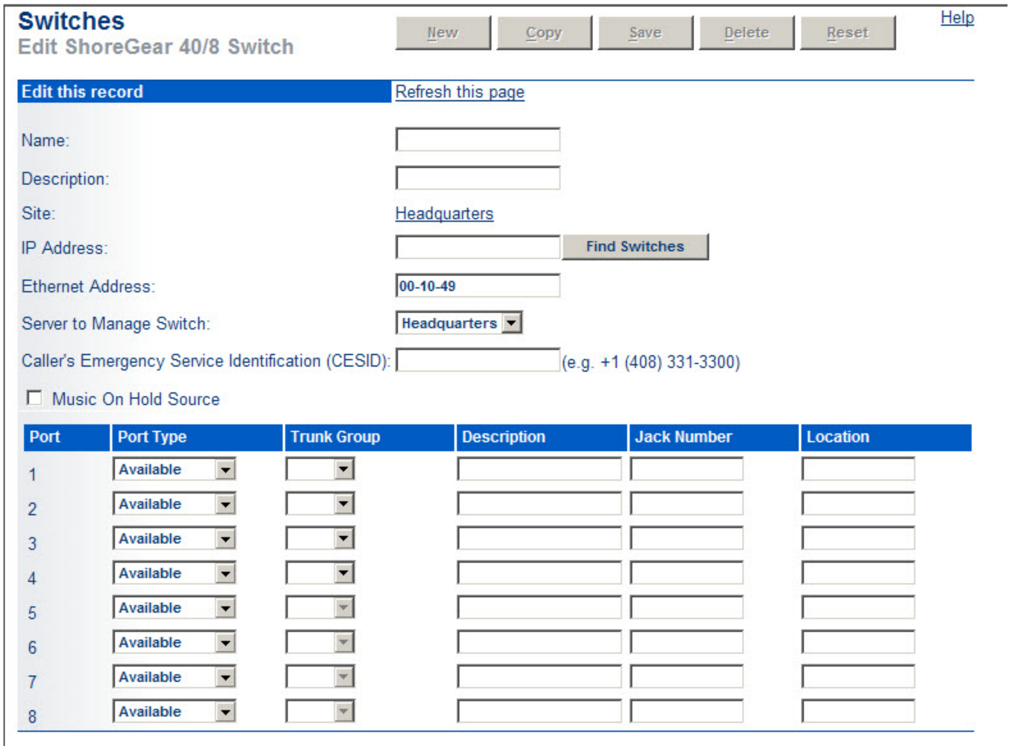


Figure 18-1 Switches Page

Step 4 In the Port Type field for the port that you want to use for SIP trunks, select the SIP trunk configuration to use:

- 5 SIP Trunks—Configures port to support up to five SIP trunk sessions concurrently.
- 100 SIP Proxy—Configures port to support up to 100 SIP proxy sessions concurrently.

Step 5 Click the **Save** button.

18.1.3.2 Creating a SIP Trunk Group

To create a SIP trunk group, do the following:

Step 1 Launch **ShoreTel Director**.

Step 2 Click **Administration > Trunks > Trunk Groups**. The Trunk Groups page appears.

Step 3 Do the following:

Step a In the “Add new trunk group at site” field, select the site for which you want to create the SIP trunk group.

Step b In to “of type” field, select **SIP**.

Step c Click **Go**. The Edit Trunk Group page appears as shown in Figure 18-2.

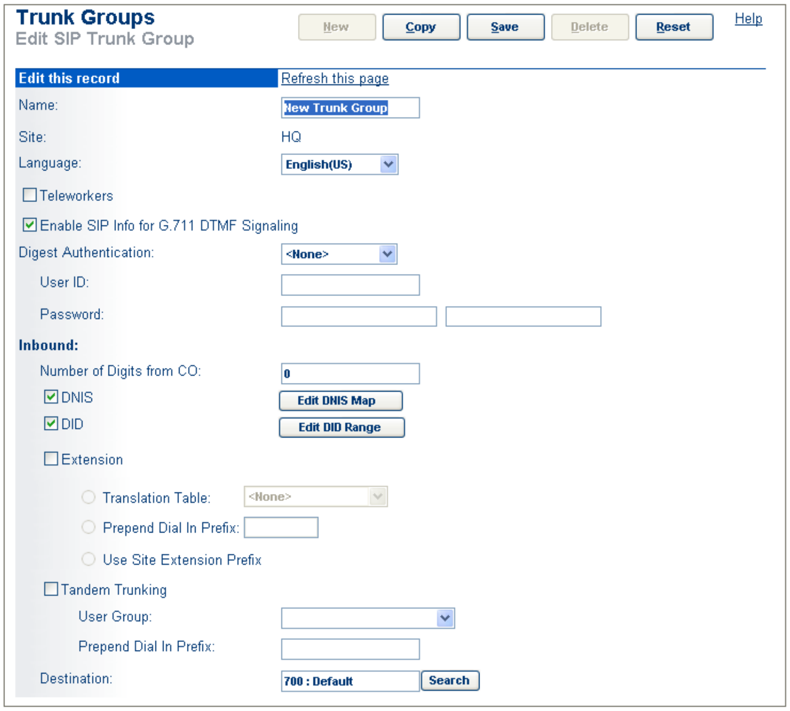


Figure 18-2 Edit SIP Trunk Group Page: Sip Specific Parameters

Step 4 Do the following to set SIP specific parameters for the trunk group:

Step a In the **Name** field, enter the name that you want to use for the trunk group.

Step b In the Language field, select the language that you want to use for this trunk group.

NOTE The Teleworkers SIP trunk parameter is not used. When teleworkers are routed through a SIP trunk:

- Audio proxies via the SG vs. the RTP are directed to ShoreTel Director or DVS.
- RTP audio packets are sent in 20 ms audio samplings instead of 10 ms.
- The inter-site call codec is used.

Step c Check the **Enable SIP Info for G.711 DTMF Signaling** check box to have SIP information sent between the SIP device and voice mail. Enable this if connecting with a ShoreTel system running a release previous to ShoreTel 8.1. Clear if the trunk is primarily used to connect a third-party SIP device.

Step d In the Profile field, select the trunk profile that you want to use for this trunk group. The options are:

- **ATTBVOIP**—Select for use with VOIP service from AT&T.
- **SystemTrunk**—Select for use with another VOIP carrier.
- **Verizon**—Select for using VOIP service from Verizon.

Step e In **Digest Authentication** field, select the type of calls for which you want the trunk to do authentication. The options are:

- **None—**Disable authentication.
- **Inbound-only—**Challenges credentials on incoming calls.
- **Outbound-Only—**Provides credentials when outbound calls are challenged.
- **All—**Authenticates all calls.

All third party SIP devices must have matching information in the associated fields, and the user ID and password of the device will be authenticated against the information stored in the ShoreTel system

Step f In the User ID field, enter the user ID that you want to use for authentication.

Step g In the Password field, enter the password that you want to use for the user.

Step 5 Do the following to configure the trunk to receive inbound calls:

Step a In the **Number of Digits from CO** field, enter the number digits that your telephone service provider will send.

Step b Check the **DNIS** checkbox to enable the trunk group to support DNIS numbers. Click **Edit DNIS Map** to create DNIS translation profiles.

Step c Check the **DID** checkbox to enable the trunk group to support DID numbers. Click the **Edit DID Range** button to specify the range of number the trunk group will accept from the PSTN.

Step d Check the **Extension** checkbox to allow the trunk group to modify extensions of inbound calls. The modification options are:

- **Translation Table—**Click this radio button to have the trunk group use a translation table, and in the field, select the translation table you want the trunk group to use.
- **Prepend Dial in Prefix—**Click this radio button to have the trunk group place a prefix in front of inbound numbers, and in the field, enter the prefix that you want the trunk group to use.
- **Use Site Extension Prefix—**Click this radio button to have the trunk group send the extension prefix associated with the site.

Step e Check the **Tandem Trunking** check box to enable tandem trunking support for the trunk group and do the following:

- In the **User Group** field, select the user group that you want to user the tandem trunks
- In the **Prepend Dial in Prefix** field, enter the prefix that you want the trunk group to attach to front of the number of inbound calls.

Step 6 Do the following to configure network call routing parameters for the trunk group for outbound calls. Figure 18-3 shows the bottom portion of the Edit Sip Trunk Group page.

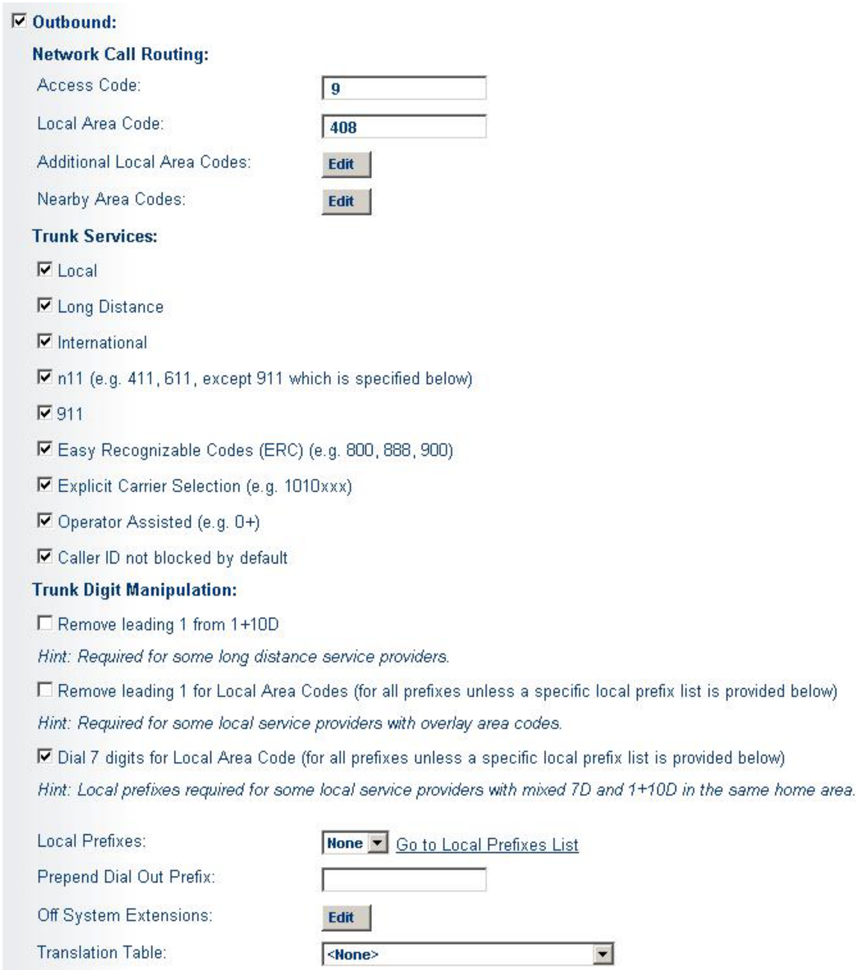


Figure 18-3 Bottom Portion of Edit SIP Trunk Group

Step a In the **Access Code** field, enter access code that you want users to dial to use this trunk. The access code structure must already be established in your dial plan. See "Setting Dial Plan Parameters" on page 35.

Step b In the **Local Area Code** field, enter the area code for location this trunk group services.

Step c Click the Addition **Local Area Code Edit** button to identify adjacent area codes to the location this trunk group services.

Step d Click the Nearby Area Code Edit button to identify area codes that this trunk group can dial to reduce toll charges.

Step 7 Do the following to configure trunk service parameters for the trunk group for outbound calls.

Step a Check the **Local** check box to enable the trunk group to place off-premise local calls.

Step b Check the **Long Distance** check box to enable the trunk group to place long-distance calls.

Step c Check the **International** check box to enable the trunk group to place international calls.

Step d Check the **Easy Recognizable Codes (ERC)** check box to enable this trunk group to support services such as toll-free dialing calls (e.g., 800, 888, 900).

Step e Check the **n11** check box to enable this trunk group to place telephone service, but not emergency, calls such as directory assistance.

Step f Check the **Emergency** check box to enable the trunk group to place emergency calls.

NOTE You must have at least one trunk group per site that allows 911 calls.

Step g Check the **Explicit Carrier Selection** check box to enable the trunk group to allow users to dial special numbers to access long-distance carriers (e.g., 1010xxx).

Step h Check the **Operator Assisted** check box to enable the trunk group to dial the outside operator (e.g., 0+).

Step i Check the **Called ID not blocked by default** check box to enable this trunk group to pass Caller ID information by default on outbound calls.

NOTE In the United States, the user can override this option with vertical service codes.

Step 8 To configure trunk digit manipulation for this trunk group, do the following:

Step a Check the **Remove leading 1 from 1+10D** check box to have the trunk drop the leading "1" your users dial on long-distance calls. (Refer to service provider to verify whether this is required.)

Step b Check the **Remove leading 1 for Local Area Codes** check box to have the trunk drop the leading "1" your users dial when dialing the local area code. (Refer to your service provider to verify whether this is necessary.)

Step c Check the **Dial 7 digits for Local Area Code** check box to enable the trunk to dial local numbers in the local area code with seven digits, if required by your local service providers.

NOTE (For all prefixes unless a specific local prefix list is provided below.)

Step d Check the Dial in E.164 Format to enable this trunk group to support E.164 numbers.

Step e In the **Local Prefixes** field, select local prefix profile that you want the trunk group to use for local calls. You can use the **Go to Local Prefixes List** link to create prefix profiles.

NOTE All prefixes that are not listed in the profile are considered long distance and calls to these numbers require a long distance trunk service.

Step f In the **Prepend Dial Out Prefix** field, enter the string that you want the trunk group to prepended outbound numbers.

NOTE A dial-out prefix is typically required when connecting to and leveraging the trunks on a legacy PBX. Dial Out Prefix is not applied to Off-System Extension calls.

Step g For **Off System Extensions**, click **Edit** to add or edit any ranges of extensions that can be accessed through this trunk group. This is typically used when setting up a tie trunk to a legacy PBX and configuring coordinated extension dialing. The Dial Out Prefix rules are not applied to Off-System Extensions.

Step h In the **Translation Table** field, select the translation table that you want this trunk group to use.

18.1.3.3 Creating a SIP Trunk

To create a SIP Trunk, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click on the **Administration > Trunks > Individual Trunks**. The Trunks by Group page appears.

Step 3 Do the following:

- In the **Add new trunk at** site field, select the ShoreTel server site in which you want to install the new SIP trunk.
- In the **in trunk group** field, select the name of the SIP trunk group with which you want to associate the new trunk.
- Click **Go**. The Edit Trunk page appears as shown in Figure 18-4.

Figure 18-4 **Edit Trunk Page**

Step 4 In the **Name** field, enter a name for the trunk.

Step 5 In the **Switch** field, select the switch with which you want to associate the new trunk.

Step 6 Select the desired SIP Trunk Type radio button. There are two choices:

- **Dynamic** - Select this radio button to provide more flexibility than a static IP address. Note that all inbound calls will be accepted, regardless of their IP address. If this is selected, you should use the authentication methods available to prevent unauthorized callers from accessing the system.

- **Use IP Address** - Select this radio button to enter a static IP address. This is recommended if the systems are static and will not be changing IP addresses often.

Step 7 In the **Number of Trunks** field, enter the desired number of SIP trunks.

Step 8 Click **Save** to store your changes.

18.1.3.4 Configure the SIP Device

SIP devices are the third-party telephones, gateways, terminal adapters, and other devices that support the protocol. The ShoreTel phones do not currently support the SIP protocol.

With each of the SIP devices you will be using, you will have to consult the manufacturer's instructions for specific instructions on configuring the device.

In a general sense, the configurations for each SIP device will be essentially the same, and will require that the following pieces of information are entered:

- IP address of the SIP server
- IP address of the SIP registrar server
- User name (identification for outbound calls)
- User information (OSE or DID)
- User password
- DTMF protocol (i.e. must support RFC 2833)

18.1.4 SIP Trunk Profiles

SIP Trunk Profiles supports custom SIP Trunk parameter settings to specify third party service provider configurations. A SIP Trunk Profile is assigned to a SIP Trunk Group from the Edit SIP Trunk Group page. See "Add or Edit a Trunk Group" on page 156 for information about adding a SIP trunk.

In addition to a generic profile, ShoreTel provides a profile that supports ATT BVOIP and Verizon.

18.1.4.1 SIP Trunk Profile List

The SIP Trunk Profiles List page, shown in Figure 18-5, displays the names of all SIP Trunk Profiles on the system. To access the SIP Extension Profile list page, select Administration -> Trunks -> SIP Profiles from the Director menu.

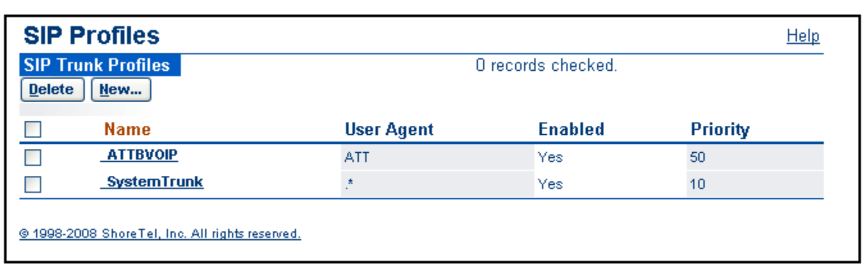


Figure 18-5 SIP Trunk Profiles List Page

Each row corresponds to a SIP Extension Profile. Table columns list the following profile parameters:

- **Name**: This parameter is the label by which Director refers to the profile.
- **User Agent**: This parameter is the expression ShoreTel uses to identify devices covered by the profile. ShoreTel compares this expression to the User Agent field in the header of SIP packets handled by the system.
- **Enabled**: This parameter lists the status of the profile. ShoreTel uses only profiles that are enabled when evaluating the characteristic set of a device.
- **Priority**: This parameter determines the order by which the profiles are evaluated against the SIP packet header. Profiles with larger priority values are evaluated before profiles with smaller priority values.

  The User Agent field of successive profiles are compared to the User-Agent field of the SIP packet header until a match is found. The profile containing the matching User-Agent field is then used to specify device configuration settings.

To add a profile, press the New button in the top left corner of the page.

To remove a profile, select the checkbox that corresponds to the profile to be deleted, then press the Delete button in the upper left corner of the page. Predefined profiles cannot be deleted.

To edit a profile, open the Edit SIP Extension Profile page by clicking the name of the profile to be modified.

18.1.4.2 Edit SIP Trunk Profile

The Edit SIP Trunk Profiles page, shown in Figure 18-6, is the Director list page from which parameter settings for the specified profile are configured. The Edit SIP Trunk Profile page is accessed from the SIP Trunk Profile List by adding a new profile or editing an existing profile. All parameter fields can be edited for User Defined profiles. Enable is the only parameter that can be modified on predefined pages.

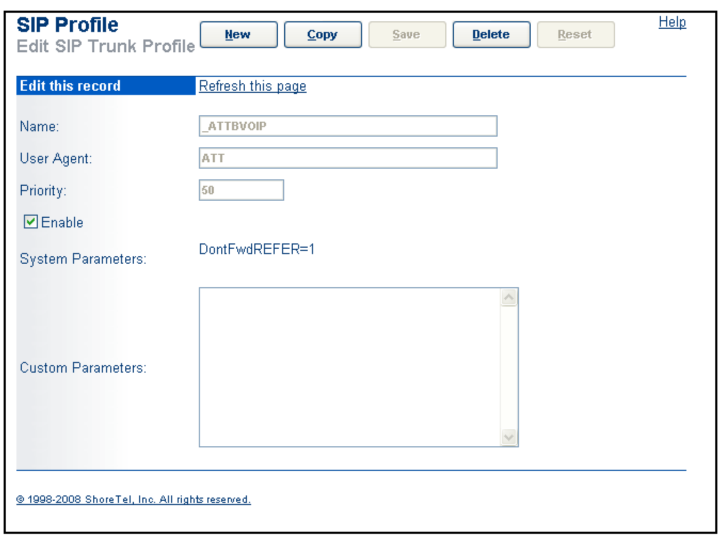


Figure 18-6 Edit SIP Extension Profile Page

Parameters listed on the page include:

- **Name:** This parameter is the label by which Director refers to the profile.

  The field contents are not editable for predefined profiles.
- **User Agent**: This parameter is the index key that ShoreTel compares to the User-Agent field in the header of inbound REGISTER or INVITE methods handled by the system.

  The field contents are not editable for predefined profiles.
- **Priority:** This parameter is used to determine the profile that is used when the User Agent field of multiple profiles match the User-Agent field in the SIP packet header.

  The field contents are not editable for predefined profiles.
- **Enable:** This parameter lists the status of the profile. ShoreTel uses only profiles that are enabled when evaluating the characteristic set of a device.
- **System Parameters:** This field lists the device characteristics and default settings.
- **Custom Parameters:** The contents of this field list additional device settings or overwrite default settings listed in the System Parameters field. Supported parameters include:
  — **DontFwdREFER**: When enabled, packets are inhibited from sending REFER messages.

18.2 Setting Up SIP Extensions

ShoreTel supports the use of SIP devices that comply with RFC 3261 (Session Initiation Protocol) as ShoreTel extensions. SIP devices configured as ShoreTel extensions support most features offered by ShoreTel IP Phones.

The ShoreTel implementation of SIP devices support standards specified by the following documents:

- RFC 3261 Session Initiation protocol (SIP)
- RFC 2833 In-band DTMF/Out-of-band DTMF
- RFC 3515 SIP Refer Method
- RFC 3550 A Transport protocol for real-time (RTP/RTCP) applications
- RFC 3551 RTP Profile for Audio and Video Conferences with Minimal Control
- RFC 2327 Session Description Protocol (SDP)
- RFC 2396 Uniform Resource Identifiers (URIs)
- RFC 2806 URLS for Telephone Calls
- RFC 3966 URIs for Telephone Calls
- RFC 2976 SIP Info Method
- RFC 3264 SDP Offer/answer
- RFC 3842 Message waiting indication reception
- RFC 3265 Subscription for MWI events

- RFC 3891 The Session Initiation Protocol (SIP) “Replaces” Header
- RFC 4208 Session Timers in the Session Initiation Protocol

18.2.1 Network Elements

Implementing SIP extensions require the following components:

- A SIP Proxy or endpoint is the logical entity to which SIP transactions communicate through a SIP device.

  ShoreTel resources can be reallocated from supporting trunks, analog extensions, or IP devices to support SIP endpoints. Switch resources are reallocated to support the following number of endpoints:

  — One trunk resource is reallocated to support 100 SIP endpoints.
  — One analog extension resource is reallocated to support 100 SIP endpoints.
  — One IP phone resource is reallocated to support 20 SIP endpoints.

  ShoreTel half-width switches also provide built-in SIP proxy resources that do not require the reallocation of other resources.

- Proxy servers are network devices that facilitate communication between SIP endpoints. The Proxy Server forwards requests from endpoints to the correct endpoint or another server.

  Within a ShoreTel network, Proxy Server functionality is built into ShoreTel half width and full width switches. ShoreTel switches provide Proxy Servers for SIP devices within the same site as the switch. Administrators can designate up to two Proxy Switches per site: one switch is assigned as the primary Proxy Server while the other switch acts as the backup Proxy Server.

  One Virtual IP address (VIP) is designated as the SIP Proxy Server address for each site that supports SIP extensions. This address must be static and is assigned to the ShoreTel switch that is configured as the SIP proxy for the site. The VIP is moved to the backup Proxy Server switch if the primary Proxy Switch fails.

  If only one Proxy Server is configured, a VIP address does not need to be configured.

- Registrars handle SIP REGISTER requests. Users send REGISTER requests for uploading their current location, which is then used by the Proxy server when routing information to the user.

  The switch that acts as the Proxy Server also serves as the Registrar for the site.

- SIP call managers routes SIP calls to and from SIP endpoints. All switches acting as SIP Proxy Servers can also perform call management functions.

  ShoreTel automatically designates a SIP endpoints call manager when it registers. SIP endpoints send register requests to the ShoreTel switch serving as its proxy server. When an unprovisioned device registers, the switch forwards the request to ShoreTel server IPCS, where it is authenticated. IPCS then assigns a call manager switch by load balancing across the site’s switch resources. If the site does not have sufficient call manager resources, the SIP device is not provisioned.

  When a call is made to or from an extension on the SIP device, the requests are sent to the call manager switch, which determines how the call is routed.

18.2.2 Supporting SIP Devices

18.2.2.1 Provisioning SIP Phones

The procedures for provisioning SIP extensions is similar to that for MGCP extensions:

1. The phone sends a register request to the registrar server.

   Existing registrations: The registrar server relays the request to SIP call manager switch, which then completes authentication.

   New registrations: The request is relayed to IPCS, which then provisions the phone on a ShoreTel switch call manager.

2. The user inputs their ShoreTel username, SIP password, and SIP proxy address (VIP address) in the configuration form provided by the phone.

   ShoreTel can recognize the extension, DID number, or the director log-in name as the user portion of a SIP address of record. The DID number is the preferred choice for creating the address of record.

3. If the device provides a URN, the system uses it to identify the device. If the device does not provide a URN, the system identifies the device by using the contact header information.

   Changing the IP address of a SIP device may result in the listing of the device by two switches. This dual listing persists until the earlier registration expires. Therefore, identifying a device by the contact header may result in a dual switch listing if the contact header changes when the phone is provisioned.

All requests to the phone are routed through the SIP call manager switch directly to the phone or to the destination specified by the active call handling mode.

All requests from the phone are sent to the proxy server, which forwards them to the SIP call manager switch.

18.2.2.2 Phone Profiles

Various SIP device models support differing sets of features, including call control capabilities, codec compatibility, and provisioning procedures. SIP Extension Profiles distinguish between the different SIP Device models, as described in the "SIP Extension Profiles" on page 527.

18.2.2.3 Phone Management

The SIP Protocol specifies phone management levels to denote the degree to which the system maintains control over the configuration of the device. Unmanaged devices do not have their configuration managed by the phone system. Managed devices have their configuration tightly managed by the system.

ShoreTel only supports unmanaged SIP devices.

18.2.2.4 Extension Assignment

Extension Assignment is a ShoreTel feature that assigns a ShoreTel extension to a ShoreTel IP Phone or a calling device external to the system. Users transfer their extension to the phones upon which they perform the Extension Assignment transfer. The device from which the extension is transferred is designated as an Anonymous Telephone if it is a ShoreTel IP Phone device.

When a user transfers an extension from a SIP device, ShoreTel recognizes the SIP device as an anonymous phone until the most recent register message sent by the phone expires. When the REGISTER message expires, the SIP device is removed from the IP phone list. While the SIP device is an Anonymous Telephone, the credentials of the user previously assigned to the device remain active.

An extension can be assigned to a SIP device through Extension Assignment only after the device becomes an Anonymous Telephone from an earlier Extension Assignment transfer and before the device is removed from the IP phone list. The only user that can use Extension Assignment to assign an extension to the device is the user that previously transferred the extension from the device.

18.2.3 User Features

This section describes the user features supported by SIP extensions.

18.2.3.1 Call Dialing and Initiation

SIP extensions support the following call initiation features:

- **Make Call:** Calls can be made from the phone or from Communicator.
- **On hook dialing:** On hook dialing is supported from Communicator.
- **Intercom:** Intercom calls are initiated from SIP devices by dialing *15 followed by #.

  Intercom calls to SIP extensions are presented as regular calls.
- **Redial and Speed dial:** Redial and Speed dial initiated from SIP extensions through Communicator operate similar to on hook dialing.

  Redial and Speed dial methods differ for each SIP device model. Feature keys on specific SIP devices may be programmed to support speed dial.
- **E911:** Calls to emergency number 911 from SIP extension sends the user's emergency identification or CESID number with the call.

  Unregistered SIP phones cannot dial 911 calls.
- **Dial plans and extension lengths:** SIP extensions support all extension lengths ShoreTel can configure. When the SIP call manager receives an Incomplete number or an illegally formed number from a SIP device, it returns the SIP response 484 Address Incomplete or 420 Bad Address, respectively.
- **Night bell:** SIP extensions can ring the night bell by dialing star code *14.

18.2.3.2 Call Handling

Call handling operations provide options for answering or routing incoming calls. SIP extensions support the following call handling options:

- **Answer call:** SIP extensions can answer calls only from the phone.

  Offering calls can be redirected to Voice Mail, an Automated Attendant, or another extension through Communicator.
- **Hang-up:** SIP extensions can hang up calls from the phone or from Communicator.
- **Ring No Answer (RNA):** The number of rings that trigger a No Answer response is specified in the Call Handling Mode definition for each user. When the No Answer

condition is triggered, the SIP call manager redirects the call to the RNA destination as specified by Director.

- **Busy:** If the user call stack size is smaller than the phone call stack size, calls that overflow the user call stack are redirected to the busy destination as specified by Director.

  When the user call stack is larger than the phone call stack, the SIP phone reject overflow calls with SIP response 486 Busy. The switch then redirects the call to the busy destination as specified by Director.

- **Forward Always:** SIP extensions supports Forward Always. When this parameter is set, all calls will be forwarded to the destination specified by Director.
- **Call waiting:** The specific call waiting implementation differs for each SIP phone model. SIP extensions support call waiting to one or multiple simultaneously offered calls for SIP devices that support this feature.
- **Call rejection:** When the SIP phone rejects the call with ***603 Decline*** response code, the switch fails the call and play the reorder tone to the phone
- **Call redirect:** If the SIP phone returns ***3xx*** response code, the switch redirects the call to the user's RNA destination. If RNA destination is not configured, the reorder tone is played.
- **Find Me:** SIP extensions support FindMe and Call Message notification.

18.2.3.3 Caller ID

Caller ID is the caller information transmitted to the other party during a voice call.

- **Caller ID presentation:** SIP extensions can display caller name and number.
- **Caller ID blocking:** SIP extensions support CID blocking and Make Number Private.
- **Caller ID for Workgroup and Hunt Group agents:** The system sends the name of the Hunt Group or Workgroup with the original caller number while the call rings. After the call is answered, the system sends the original caller name and number.

18.2.3.4 Call Control

Users perform call control operations to manage their active voice calls. SIP extensions support the following call control operations:

- **Hold:** Call hold and unhold are performed from the phone. Implementation of the reminder ring for held calls differs among SIP phone models. This feature is handled internally by the phone.
- **Basic transfer:** SIP extensions support blind transfers that use REFER. Transfers using re-INVITE are not supported.
- **Consultative transfer:** SIP extensions support consultative transfers that use REFER. Transfers using re-INVITE are not supported.
- **Park from SIP phone:** Calls are parked from SIP extensions by dialing *11 followed by #.
- **Unpark on SIP phone:** SIP phone users pickup calls parked from their extension stack by dialing *12 followed by their extension number.

Taking the SIP phone offhook is not sufficient to resume a call parked on the same extension.

- **Pickup:** SIP extensions pickup calls by dialing *13 followed by #.
- **Unpark:** SIP extensions unpark calls by dialing *12 followed by #.
- **Conference Calls:** Three party conference calls initiated from the phone use the phone's resident MCU.
  - — Three party conferences calls initiated from Communicator uses MakeMe conferencing.
  - — MakeMe conferencing is used when a SIP phone joins a conference call.
  - — Four to six party conference calls are supported using MakeMe and must be initiated through Communicator.
- **Call recording:** SIP Extension call recordings are supported only for only calls involving a physical trunk and can be initiated only through Communicator.
- **Voicemail:** SIP extensions reach voice mail by dialing # on the phone or pressing the Communicator VM button.
- **MWI:** SIP extensions support Message Warning Indicator by using NOTIFY or SUBSCRIBE/NOTIFY on phone models that support MWI. MWI is configurable through SIP phone profiles.
- **Agents:** SIP extensions are available for Workgroup and Hunt Group agents.
- **Bridged call appearance:** SIP extensions do not support Bridged Call Appearances.
- **DTMF:** SIP Extensions support DTMF tones as specified by RFC 2833.
- **Huntgroup busyout:** SIP extensions can busyout huntgroups by dialing *18 followed by HG#.

18.2.3.5 Specialty Calls

Specialty calls establish communication between system users during voice calls.

- **Silent monitor:** SIP Extensions cannot initiate this operation. Silent Monitor cannot be performed on SIP extension phones.
- **Barge In:** SIP Extensions cannot initiate this operation. Barge In cannot be performed on SIP extension phones.
- **Whisper Page:** SIP Extensions cannot initiate this operation. Silent Monitor cannot be performed on SIP extension phones.

18.2.4 System Features

SIP extensions support the following system features:

- **Account codes:** Users on SIP extension can be forced to use account codes for external calls.
- **Bandwidth allocation:** ShoreTel allocates bandwidth for SIP voice calls.
- **Backup Auto Attendant (BAA):** The SIP call manager switch provides BAA to the SIP extension. All BAA prompts are played in G.711 format.

- **Supported Codecs:** ShoreTel default settings support negotiation of the following codecs:
  - — L16/16000
  - — L16/8000
  - — AAC_LC/32000
  - — PCMU/8000
  - — PCMA/8000
  - — G722/8000
  - — DVI4/8000
  - — BV32/16000
  - — BV16/8000
  - — G729/8000

  New codecs can be added to the Supported Codec List. This list is accessed from Director by selecting **Administration > Call Control > Supported Codecs** from the main menu.
- **Fax redirection:** Fax calls to SIP extensions are redirected to the site fax redirect number. SIP extensions do not support T.38.
- **Music on Hold (MoH):** ShoreTel does not provide music on hold to SIP extensions
- **Extension Assignment – external devices:** SIP extension user can be configured such that Extension Assignment can be performed from devices external to the ShoreTel system.
- **On-Net dialing:** SIP extensions can use on-net dialing.
- **PSTN failover:** SIP extensions can be reached via PSTN failover.
- **Packetization period:** The default packetization period for all calls involving SIP extensions is 20 ms.
- **Video call:** The "Allow intersite video calls" setting restricts video calls across sites. ShoreTel does not allocate bandwidth for video calls.
- **Country Call Progress Tones and Ring Tones:** SIP extensions provide call progress and ring tones for countries supported by ShoreTel.
- **Language support:** SIP extensions provide support for languages as required by the countries supported by ShoreTel.

18.2.5 User Assignment

ShoreTel can use the extension, DID, or Client User ID to identify or authenticate a user.

The contact name, or the information that is placed in the SIP header for messages from the device, is constructed from the contact name configured in the SIP device.

The address of record is the IP address by which a user can be reached through the SIP device. The address of record is from the user name configured in the SIP device.

18.2.6 Configuration

This section describes the procedures required to implement SIP extensions.

18.2.6.1 Specifying the SIP Network Elements

The Edit Sites page supports the following SIP Network configuration tasks for the specified site:

- Defines the IP address for the SIP Proxy and Registrar Server.
- Designates the ShoreTel switches that serve as the site's SIP Proxy and Registrar servers. Data entry fields are available for designating primary servers and secondary servers.

To configure the SIP Network Elements for a ShoreTel site:

Step 1 Launch ShoreTel Director.

Step 2 Select **Administration** > **Sites**. The Sites page appears.

Step 3 Click the site that you want to edit. The Edit Site page appears as shown in Figure 18-7.

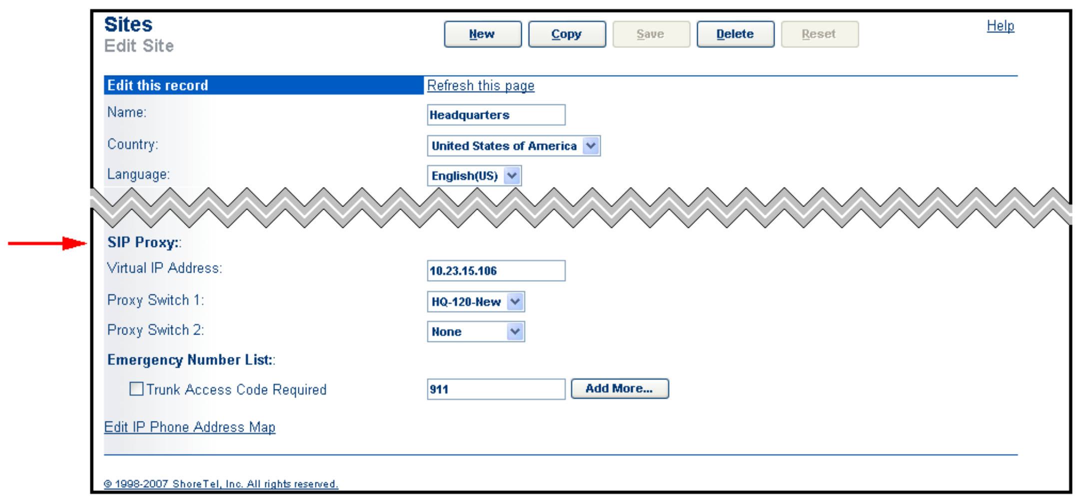


Figure 18-7 Configuring SIP Network Elements on the Edit Site Page

The SIP Proxy section, located in the bottom section of the page, configures the following parameter settings:

- **Virtual IP Address:** This parameter defines the IP address of the site's SIP Proxy Server and Registrar server. The IP address is independent of the switch that performs the server functions. SIP extensions require that this parameter is set to a valid address.
- **Proxy Switch 1:** This setting designates the switch that performs the site's SIP server functions. The drop down menu lists all switches assigned to the site. SIP extensions require the setting of this parameter.
- **Proxy Switch 2:** This setting designates the switch that performs the site's SIP server functions when the switch specified by Proxy Switch 1 is not available. This parameter is optional.

18.2.6.2 Setting the SIP Call Controls

The Edit Call Control Options panel configure SIP parameters for the ShoreTel system.

To configure the SIP Network Elements for a ShoreTel Voice Switch:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration -> Call Control -> Options**. The Edit Call Control Options page appears as shown in Figure 18-8.

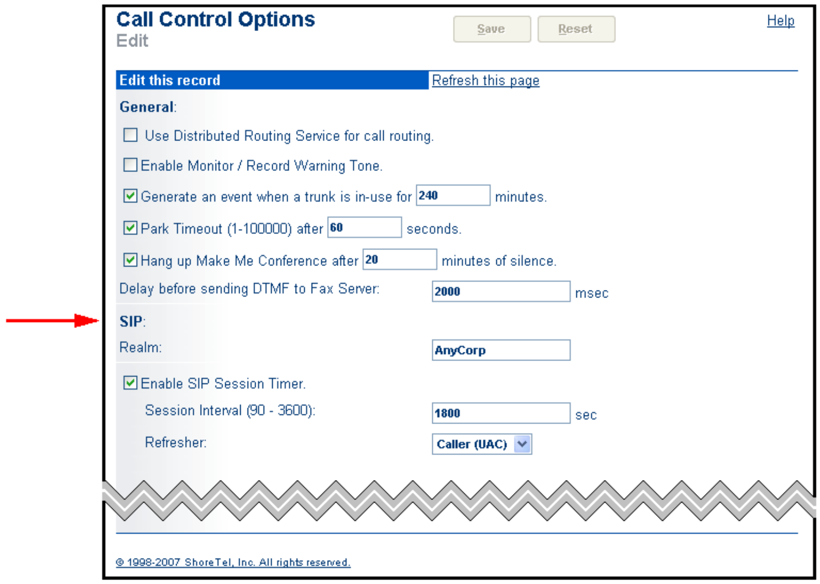


Figure 18-8 Editing SIP Parameters in the Call Control Options Page

Step 3 Set values for the following call control parameters:

- **Realm:** A realm is a protection domain over which SIP authentication parameters are specified. For digest authentication, each such domain defines a set of usernames and passwords, through which access is granted. The Realm parameter configures the name of the realm to which the site devices belong.

  Resetting the Realm requires a system reboot.

- **Enable SIP Session Timer:** SIP Session Timer measures the timeout period that determines when devices transmit or receive a RE-INVITE or UPDATE method to refresh a session still in progress. This timeout informs stateful proxies that a call remains active.

  When Enable SIP Session Timer is selected, the following parameters are active:

  — **Session Interval:** This parameter specifies the keep alive period for SIP Sessions when Enable SIP Session Timer is enabled.

  — **Refresher:** This parameter designates the device – the UAC or UAS – that sends the refresh. The method (RE-INVITE or UPDATE) is dynamically selected based on the Timer header supported by the SIP endpoint.

18.2.6.3 SIP Extension Profiles

A SIP Extension ShoreTelProfile is a ShoreTel record, defined for a specific SIP device, that lists characteristics, properties, features, and settings for that device. A ShoreTel Voice Switch uses SIP Extension Profiles to monitor and service the SIP devices connected to the system. ShoreTel supports **predefined** and **user defined** profiles.

- **Predefined profiles** are provided by ShoreTel to support generic devices or devices for which a specific profile is not defined. Although predefined profiles cannot be deleted or modified, they can be deactivated or superseded by user defined profiles.
- **User defined profiles** are created through Director and list parameter settings for specific SIP device models.

SIP Extension Profile List

The SIP Extension Profiles List page displays the names of all SIP Extension Profiles on the system. To access the SIP Extension Profile list page, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > IP Phones > SIP Profiles**. The Sip Profiles page appears as shown in Figure 18-9.

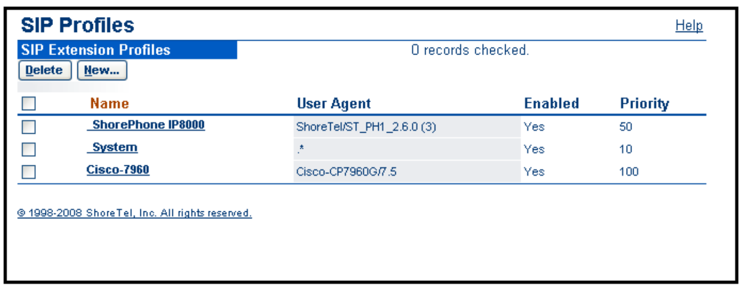


Figure 18-9 SIP Extension Profiles List Page

Each row corresponds to a SIP Extension Profile. Table columns list the following profile parameters:

- **Name**: This parameter is the label by which Director refers to the profile.
- **User Agent**: This parameter is the expression ShoreTel uses to identify devices covered by the profile. ShoreTel compares this expression to the User Agent field in the header of SIP packets handled by the system.
- **Enabled**: This parameter lists the status of the profile. ShoreTel uses only profiles that are enabled when evaluating the characteristic set of a device.
- **Priority**: This parameter determines the order by which the profiles are evaluated against the SIP packet header. Profiles with larger priority values are evaluated before profiles with smaller priority values.

  The User Agent field of successive profiles are compared to the User-Agent field of the SIP packet header until a match is found. The profile containing the matching User-Agent field is then used to specify device configuration settings.

To add a profile, press the New button in the top left corner of the page.

To remove a profile, select the checkbox that corresponds to the profile to be deleted, then press the Delete button in the upper left corner of the page. Predefined profiles cannot be deleted.

To edit a profile, open the Edit SIP Extension Profile page by clicking the name of the profile to be modified.

Edit SIP Extension Profile

The Edit SIP Extension Profiles page, shown in Figure 18-10, is the Director list page from which parameter settings for the specified profile are configured. The Edit SIP Extension Profile page is accessed from the SIP Extension Profile List by adding a new profile or editing an existing profile. All parameter fields can be edited for User Defined profiles. Enable is the only parameter that can be modified on predefined pages.

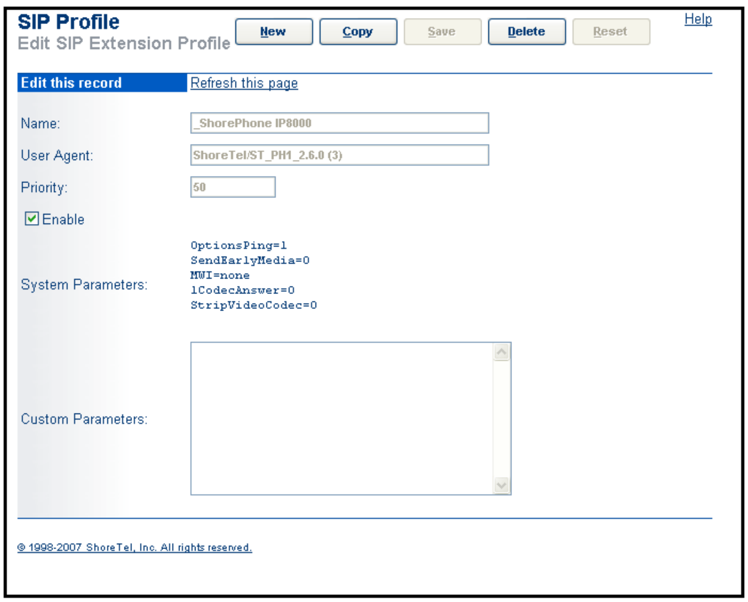


Figure 18-10 Edit SIP Extension Profile Page

Parameters listed on the page include:

- **Name:** This parameter is the label by which Director refers to the profile.

  The field contents are not editable for predefined profiles.

- **User Agent:** This parameter is the index key that ShoreTel compares to the User-Agent field in the header of inbound REGISTER or INVITE methods handled by the system.

  The field contents are not editable for predefined profiles.

- **Priority:** This parameter is used to determine the profile that is used when the User Agent field of multiple profiles match the User-Agent field in the SIP packet header.

  The field contents are not editable for predefined profiles.

- **Enable:** This parameter lists the status of the profile. ShoreTel uses only profiles that are enabled when evaluating the characteristic set of a device.

- **System Parameters:** This field lists the device characteristics and default settings.

- **Custom Parameters:** The contents of this field list additional device settings or overwrite default settings listed in the System Parameters field. Supported parameters include:
  - **optionsping:** Set this parameter to 1 if the SIP device can process SIP OPTIONS command. This command is sent to the SIP device as Keepalive message.
  - **sendearlymedia:** Set this parameter is 1 when the SIP device is capable of receiving "early media". Some BAA prompts are streamed as early media.
  - **mwi:**
    - Set to *none* if the SIP device does not support MWI
    - Set to *subscribe* when the SIP device subscribes for message waiting service.
    - Set to *notify* when the SIP device will can receive MWI notification without subscribing for the service.
  - **1CodecAnswer:** When set to 1, only one codec is set in answer SDP.
  - **StripVideoCodec:** When set to 1, ShoreSIP UA strips video codec from SIP SDP.
  - **AddGracePeriod:** When extra time needs to be added to the expire time for SIP registrations
  - **FakeDeclineAsRedirect:** When set to 1, response code "603 decline from SIP endpoint" is treated as redirect to CHM destination.

18.2.6.4 Allocating Switch Resources

A SIP extension can be utilized only if an unused SIP Proxy Port is available on a switch assigned to the site. Proxy ports are allocated through the Edit Switch page. Figure 18-11 displays the Edit ShoreTel 50 Switch page.

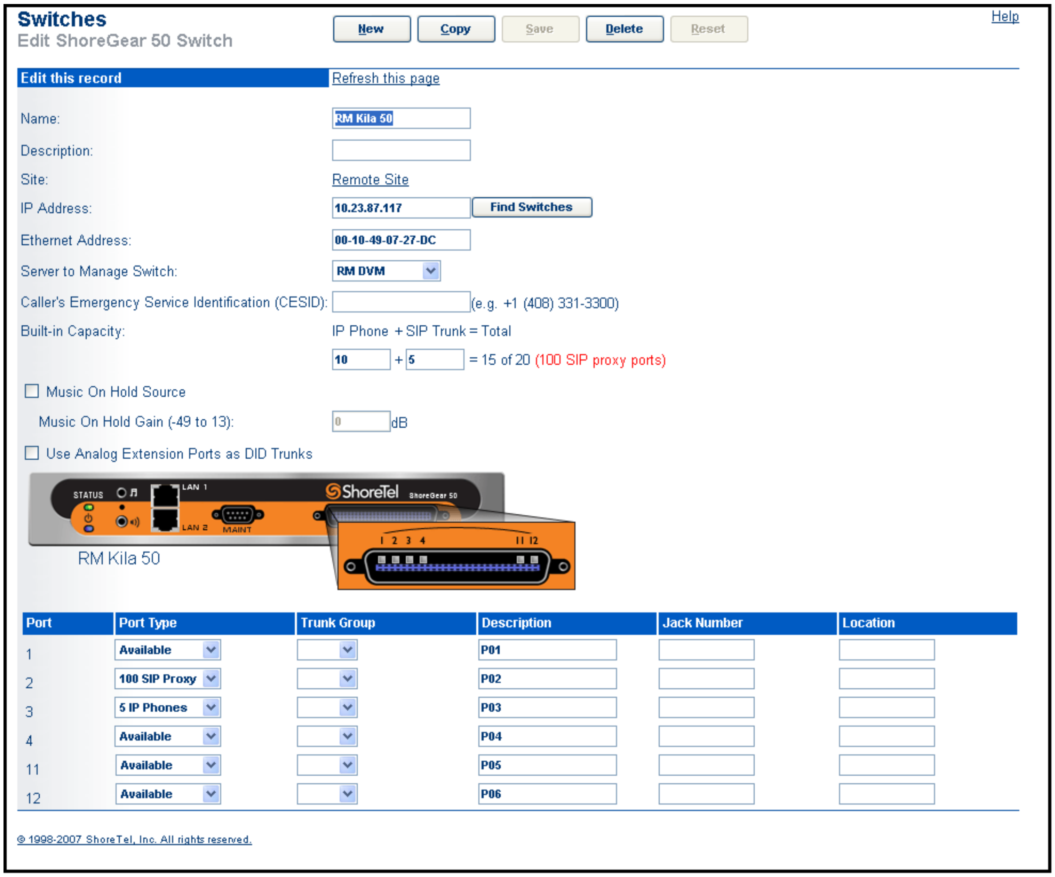


Figure 18-11 Allocating SIP Proxy Ports

ShoreTel switches provide two SIP proxy port sources: Built-in capacity and Port assignment.

- **Built-in capacity:** ShoreTel half-width switches, such as the ShoreTel Voice Switch-50, provide IP phone, SIP trunk, and SIP proxy resources that are independent of port switches. The number of resources varies with each switch model. Each resource unit supports one IP phone, one SIP trunk, or five SIP Proxies.

  The example in the figure indicates that the SG-50 provides 20 resource units. The resources allocation configure is:

  — 10 resources as IP phones
  — 5 resources as SIP trunks
  — 5 resources that provide 100 SIP proxy ports.

To configure the Built-in resource for SIP Proxy Port allocation, enter the desired number of IP Phone and SIP Trunk resources in the specified data entry fields. The remaining resources from the total available are automatically designated for use as SIP proxy ports.

- **Port resources:** Switch ports can be configured to support 100 SIP proxy ports.

To configure a port for SIP Proxy Port allocation, access the drop down menu of the desired port in the port section of the page and select the 100 SIP Phones option.

18.2.6.5 Setting an Extension Password

An extension is enabled for SIP if a value is assigned to the User's SIP Password parameter. The SIP Password parameter is located at the bottom of the Edit User: General page, as shown in Figure 18-12. Clearing the SIP Password data fields disables the extension from supporting SIP.

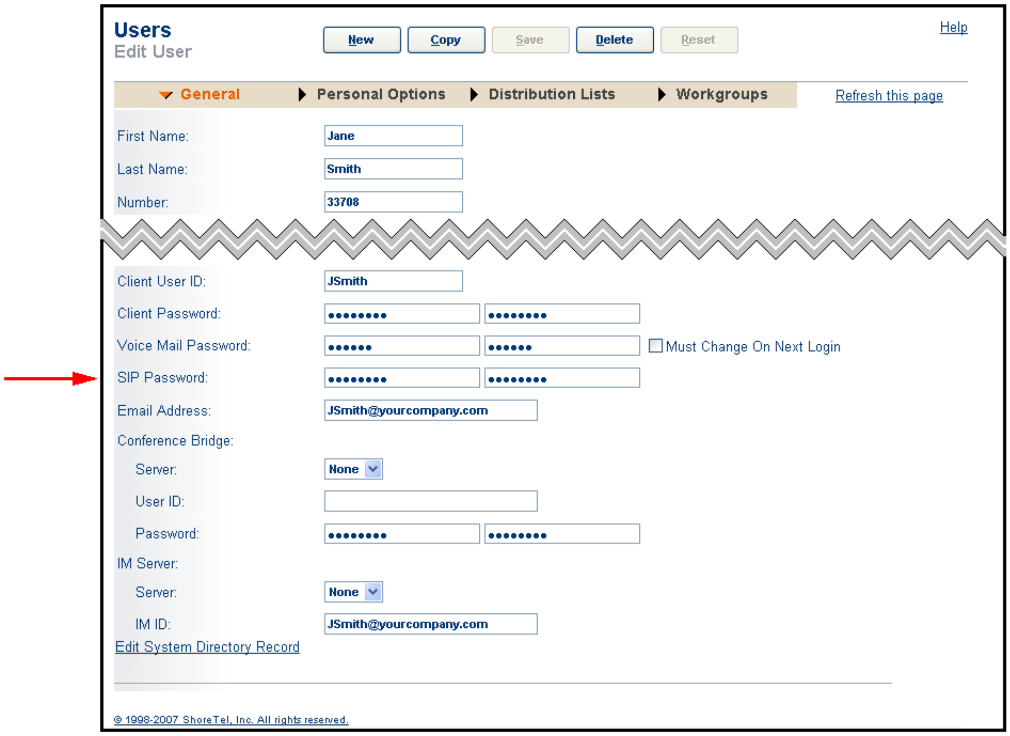


Figure 18-12 Setting a SIP Password

18.2.6.6 Monitoring SIP Phones

All SIP extension devices on the ShoreTel network are listed on the IP Phone page in Director, as shown in Figure 18-13. SIP devices are deleted from this page when they are physically removed from the network.

Clicking on the name of a SIP device displays an Edit IP Phone page for that device. The Edit IP Phone page displays, in addition to the information listed on the IP Phones list page, the Credential Name, User Name, Contact, and Address of Record for the device. The only information editable from this page is the name of the device.

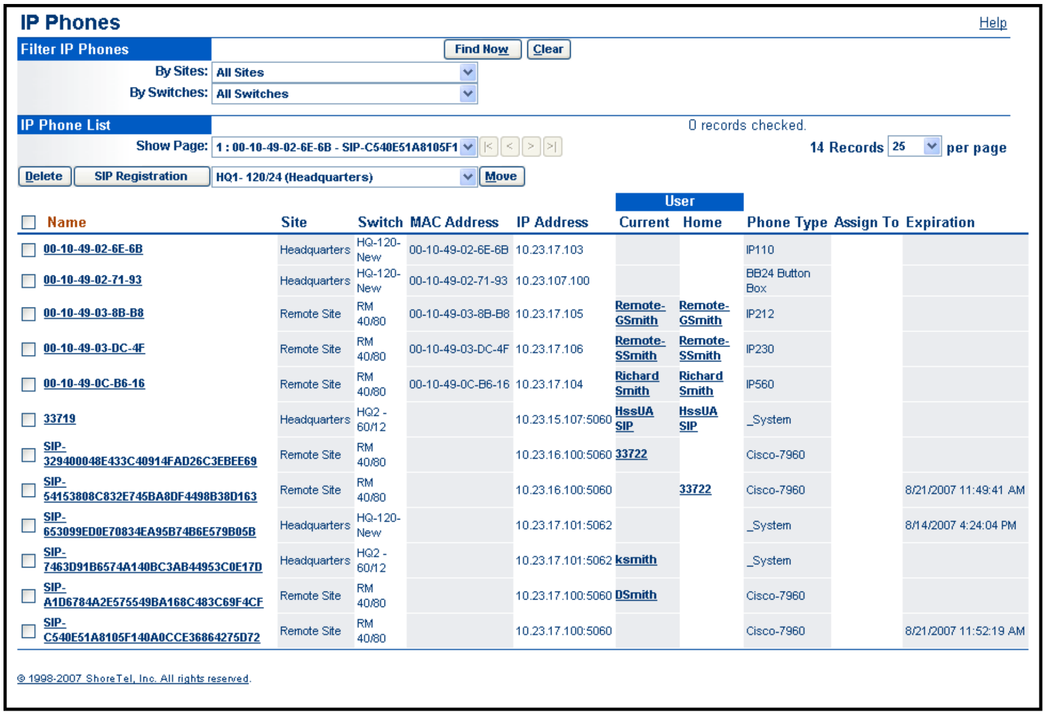


Figure 18-13 SIP devices listed on IP Phones list Page

SIP Register popup

The SIP Register popup, shown in Figure 18-14, displays registration information for the corresponding device. This popup is accessed by pressing the SIP Registration on the IP Phones page.

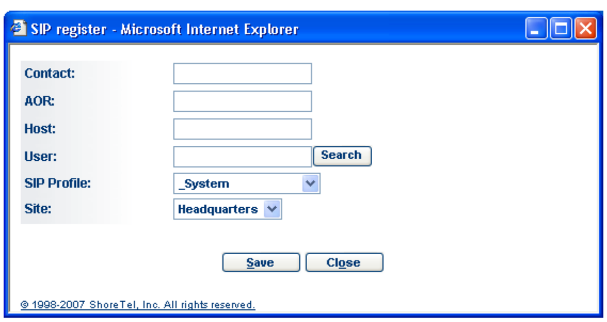


Figure 18-14 SIP Register popup Page

SIP devices can be manually registered using SIP Registration button on the IPPhones page.

18.3 Setting Up SIP Integration with Third-Party Unified Messaging Systems

ShoreTel software allows organizations to integrate with external third-party Unified Messaging (UM) systems. Enterprise organizations that want to take advantage of deploying a separate voicemail or fax server within the ShoreTel environment can do so.

After the ShoreTel solution is deployed, an organization can use third-party solutions such as Microsoft Exchange to retrieve voicemail and send faxes through any ShoreTel-supported UM fax server.

Microsoft Exchange Server 2007 and Microsoft Exchange Server 2010 are certified and supported by ShoreTel 12.1 and 12.2.

Support for integrating the ShoreTel solution with third-party UM systems gives enterprise organizations the flexibility of deploying a separate Unified Messaging (UM) server for voicemail or fax.

18.3.1 General Feature Consideration

Integration with third-party UM systems are subject to the following considerations:

- Moving ShoreTel users to a third-party UM server deletes existing voicemails on the ShoreTel system. We recommend that you save any existing voicemails before upgrading to ShoreTel.
- ShoreTel Communicator and Communicator for Mobile do not support voicemail for MSE users.
- Voicemail with Outlook integration is not available from Communicator.
- Message waiting indication is not available for ShoreTel Communicator.
- Although messaging is available from ShoreTel, the following voicemail features are not available to a user when ShoreTel is integrated with a third-party UM systems:
  - Any Phone
  - Find-Me
  - Escalation Profiles
- Switching between ShoreTel and External SIP Unified Messaging voicemail results in the following conditions.
  - Loss of all existing ShoreTel messages
  - Users might need to re-create the ShoreTel Communicator rules to reflect the new voicemail number.

18.3.2 Configuration

This section describes the procedures required to configure a ShoreTel SIP UM server.

To integrate with third-party UM system, you must first configure your ShoreTel SIP server through ShoreTel Director, and then setup and configure one of the ShoreTel supported 3rd.-party UM solutions. This section describes how to configure a ShoreTel SIP UM server.

NOTE SIP Unified Messaging Servers cannot be added to the system if Always Use 5004 Port is checked on the Call Control–>Options page. Uncheck this flag before proceeding.

NOTE Enable a ShoreTel Voice Switch to be the SIP proxy for the site where you want to add the SIP UM Server. This switch is referred to as the site proxy switch.

18.3.2.1 Configuring the SIP Server

To set up and configure a new SIP Unified Messaging server, perform the following steps.

NOTE You need a ShoreWare External Unified Messaging SIP Link license to use the “Allow Extenal Voice Mail for Extension-Only User” option.

Step 1 Launch ShoreTel Director and log in as the administrator.

Step 2 Click **Administration > SIP Servers > SIP Servers**. The SIP Servers page appears.

Step 3 Click **New**. The SIP Server Info dialog box shown in Figure 18-15 appears.

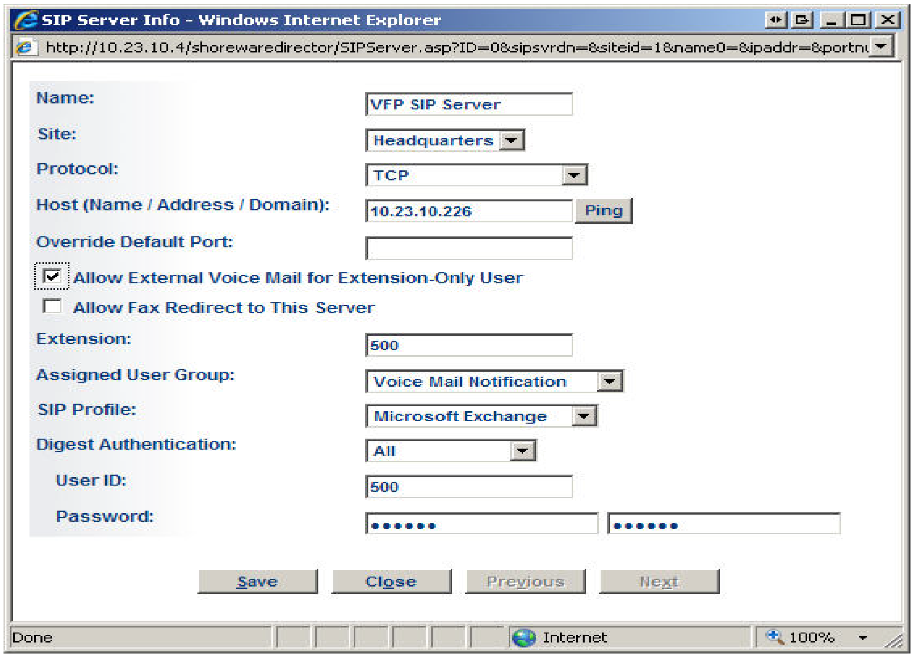


Figure 18-15 SIP Server Info Screen

NOTE: Before putting a mark in the checkbox for “Allow External Voice Mail for Extension-Only User,” the administrator should verify the presence of a license for ShoreTel External Unified Messaging SIP Link.

Step 4 Enter the SIP Server information for the new server.

Table 18-1 displays the IP Server configuration fields and descriptions.

Table 18-1 SIP Server Info Requirements

Field	**Description**
Name	Microsoft Exchange SIP Server name.
Site	Site where this UM Server will reside. Note that a popup message will occur if you change the UM Server site later.
Protocol	Select SIP transport protocol for SIP UM server. Select TCP for Microsoft TCP Microsoft Exchange server.
Host (Name/Address/ Domain):	IP address or Host name of the SIP Server.
Override Default Port	Default port is 5060. If you want to use another value, use this field.

Table 18-1 SIP Server Info Requirements (Continued)

Field	Description
Allow Fax Redirect to This Server	Enable this to use this server as Site Fax Server.
Extension	Enter a unique extension for SIP UM Server. This should be identical to the pilot number assigned on the SIP UM Server.
Assigned User Group	This is used to provide outbound trunk calling capability from the MSE server. Please assign a user group with outbound trunk access to this SIP Server. In case of Exchange server, this user group will be used to make external calls for these features: • MS outlook play on phone feature for playing voice mail on external number. • Call sender feature to call to an external user who left the voice mail.
SIP Profile	Select the right profile from the list, based on your SIP Server. If using Exchange server, select Microsoft Exchange profile.
Digest Authentication	Select from drop down, depending on what is supported by the SIP UM server. Select none for MSE.
User ID	User name for authentication, if enabled.
Password	Password for authentication, if enabled.

Step 5 Click **Save** when you are finished.

18.3.2.2 Creating a User Group to the Use SIP Server

After configuring a SIP server, you must create a user group for SIP server users.

To create a user group for SIP server users, perform the following steps.

Step 1 Launch ShoreTel Director and log in as the administrator.

Step 2 Click the **Administration > User Groups**. The User Groups page appears.

Step 3 Click the user group that you want to use SIP trunks or and click **New** create a new user group. The Edit User Group page appears.

Step 4 In the Voice Mail Interface Mode field, select **External Voice Mail**, **SIP**. A pop-up message appears.

Step 5 Click **OK**.

The CHM destinations for this user are set to the selected SIP server extension.

18.3.2.3 Configuring a User To Have SIP Server Access

To create a user within a User Group for a SIP Server, follow these steps.

Step 1 Launch ShoreTel Director and log in as the administrator.

Step 2 Click **Administration > Users > Individual Users**. The Individual User page appears.

Step 3 Select a **User** that you want to with **SIP server** permissions. The Edit User page shown in Figure 18-16 appears.

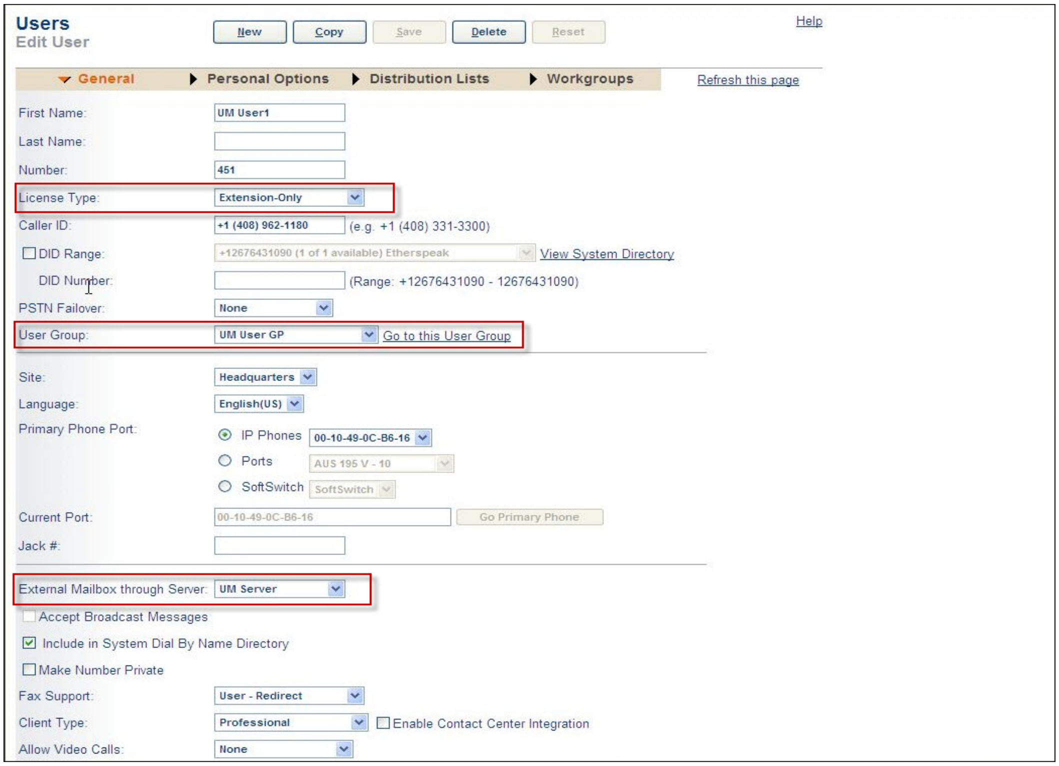


Figure 18-16 Creating a User within a User Group for a SIP Server

Step 4 In the License Type field, select **Extension Only**.

Step 5 In the User Group field, select the user group configured to use a SIP voice mail interface for the user.

Step 6 In the External Mailbox through Server **field**, select **UM Server** from the drop-down list.

NOTE The limit for simultaneous active calls is 24 per switch. The active calls are SIP calls handled by a ShoreTel Voice Switch that could be voice, T.38, or a combination of these.

18.3.2.4 Configuring a Third-party Unified Messaging Server

After you have successfully configured ShoreTel Director, you must setup and configure your third-party UM server. Please contact the ShoreTel Technology Partners Program at *http://www.shoretel.com/partners/technology/certified_partners.html* for more information on configuring ShoreTel supported third-party UM solutions.

NOTE A Unified Messaging SIP Link license is required for every Unified Messaging (SIP) server added in ShoreTel Director.

CHAPTER 19

Maintenance

The ShoreTel system provides maintenance information through ShoreTel Director to let system administrators check the operational status of switches, ports, servers, and services. It also provides information to resolve an event or error that might occur while the system is running. Operational status and events are tracked by the system. The information is displayed on the maintenance pages that you access from the Maintenance section in the navigation frame.

This chapter provides information about the maintenance functions available in ShoreTel Director. The topics include:

19.1 Quick Look

The Quick Look maintenance page provides a snapshot of the entire ShoreTel system. It includes information about each site and the corresponding switches and servers. To access the Quick Look page, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click Maintenance > Quick Look. The Quick Look page appears as shown in Figure 19-1.

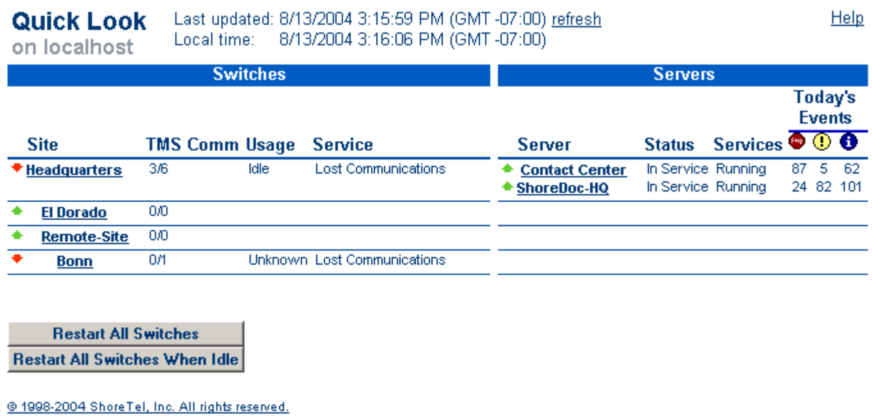


Figure 19-1 Quick Look Page

The Quick Look page is described as follows:

- **Refresh:** Click this word, located at top of the page, right of the *Last Updated* time, to update the page information. The Quick Look page automatically updates every 60 seconds.
- **Switches:** This area lists all of the sites on your system. Sites without servers are listed beneath their associated site that has a server. The following is represented under Switches:
  - — **Site***:* Clicking the name of a site will bring you to the Maintenance Switches Summary page.
  - — **TMS Comm***:* This summarizes the communication state of all switches at the site. The first number represents switches with which TMS can currently communicate. The second number is the total number of switches at the site.
  - — **Usage***:* Shows either an Idle or In Use state.
  - — **Service***:* This summarizes the service state of all the switches at the site.
- **Servers:** This area lists all of the ShoreTel servers.
  - — **Server***:* Sites that have a server list the server in bold and a link to the Quick Look Server Maintenance page is provided. Sites without servers list the associated server they are served by.
  - — **Status***:* This summarizes the status of the server.
  - — **Services***:* This summarizes the status of the ShoreTel services on the server.
  - — **Today's Events***:* This summarizes the events that have been recorded in the event log on the server. See the "Quick Look Server Events" on page 555.
- **Apply this Command to All Switches:** Each option initiates restart operations for all system switches. The restart begins immediately when an option is selected:
  - — **Restart***:* Select this option to immediately shut down and restart all switches. Calls that the switches are servicing when this command is selected are lost.
  - — **Restart when idle**: Select this option to shut down and restart, each switch once the calls they are currently servicing are completed. Calls that are in place when this command is selected are allow to completely normally.
  - — **Reboot***:* This option performs the same operation as Restart.
  - — **Reboot when idle***:* This option performs the same operation as Reboot when idle.

19.1.1 Maintenance Switches Summary

The Maintenance Switches Summary page (shown in Figure 19-2) is accessed by clicking on a site name listed on the Quick Look page. The switches summary lists all ShoreTel voice switches configured at the site. Switches are identified by name and IP address.

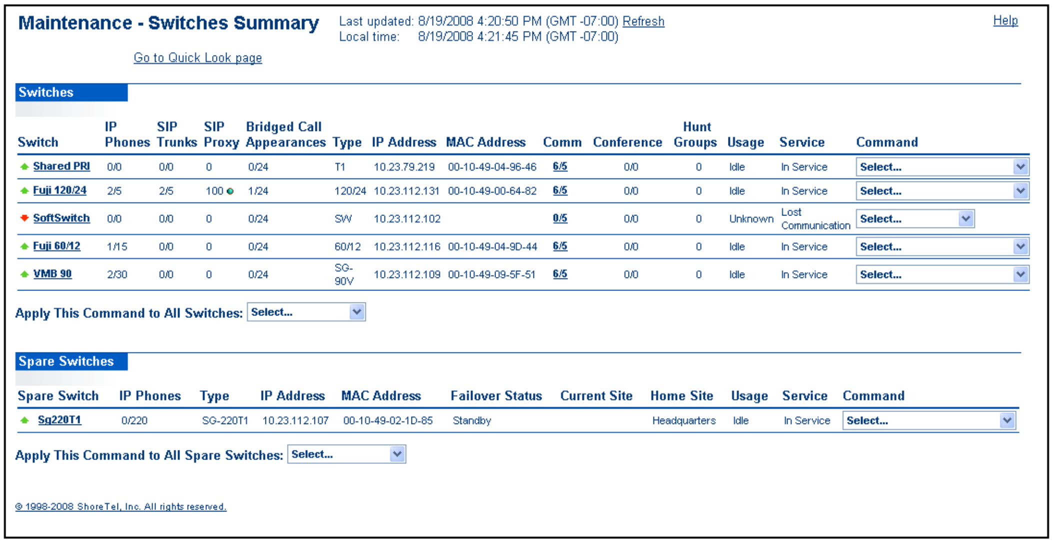


Figure 19-2 Maintenance Switches Summary Page

The Restart All and Restart All When Idle buttons restart all switches. When clicked, Restart All restarts all switches and discontinues any calls in progress. Clicking Restart All When Idle restarts each switch only after all calls have ended.

19.1.1.1 Voice Switches and Service Appliances Elements

The columns in the Voices **Switches and Service Appliance** section includes the following elements:

- **Switch/Appliance:** Lists the names of the voice switches and service appliances that are installed on the system. Clicking a name invokes the maintenance page for that device.
- **IP Phones:** Lists the number of IP phones currently registered with the device and the device capacity.
- **SIP Trunks:** Lists the number of SIP trunks currently registered with the device and the capacity of SIP trunks the device is configured to support.
- SIP Proxy:
- **BCA:** Lists the number of Bridge Call Appearance (BCA) extensions currently registered with the device and the BCA capacity the device is configured to support.
- **Type:** Lists the type of device.
- **IP Address:** Lists the IP address assigned to the device.
- **MAC Address:** Lists the Ethernet address of the device.

- **Comm:** Lists the number of devices this device with which this switch is currently communicating and the total number of devices on the network with which this device can communicate.

  NOTE Clicking the Comm fraction takes you to the Switch Connectivity page, which provides more detail on each switch's communication status.

- **Audio Ports:** Lists the number of audio conference ports the device is currently using and the number of audio conference ports the device is licensed to use.
- **Web Ports:** Lists the number of Web conference ports the device is currently using and the number of Web conference ports the device is licensed to use
- **Conference:** Indicates the number of conference ports currently in use.
- **Hunt Groups:** Lists the number of hunt groups registered with the device.
- **Usage:** This is the usage state of the device.
  - **Unknown—**The usage state is unknown. This might be true because communication between the server and the switch might have been lost.
  - **In Use—**At least one port or IP phone has an active call.
  - **Off Hook—**At least one port or IP phone is off-hook, but no ports are in use.
  - **Idle—**There are no ports or IP phones off-hook or in-use.
- **Service:** This is the service state of the device.
  - **Lost Communication—**The server lost communication with the voice switch. Note that the voice switch may be fully operational but the ShoreTel server cannot see the voice switch due to a networking issue. This also occurs when the voice switch is powered off.
  - **Upgrade in Progress—**The voice switch is currently being upgraded with a new software version.
  - **Restart Pending—**A Restart when idle command was issued but the restart did not occur because ports are still in use.
  - **Firmware Version Mismatch—**The voice switch is running a version of software that does not match the version on the server. The voice switch continues to run call control but does not have access to any voice services on the server. This typically happens on software upgrades after the server was upgraded but before the voice switches were restarted and upgraded. Note that voice switches at the same firmware version also continue to operate together.
  - **Firmware Update Available—**This means the server has a new optional version of firmware available for the voice switch. A voice switch in this state continues to run call control as well as access the voice services on the server. This state typically happens when you install a patch on the ShoreTel server. If you want, or need, the patch to propagate to the voice switches, you must restart them.
  - **FTP Booted—**This means the ShoreTel voice switch did not boot from FLASH memory but booted from an FTP server, most likely on the ShoreTel server. You can correct this problem by rebooting the voice switch. If this does not correct the problem, please contact ShoreTel's Customer Response Center.
  - **Port Out of Service—**All ports on the ShoreTel Voice Switch are out of service.
  - **IP Phone(s) Out of Service—**One or more IP phones associated with the switch are out of service.

— **Port Out of Service—**This means one or more, but not all, ports or IP phones are out of service on the ShoreTel voice switch. Ports or IP phones typically go out of service because either someone manually put them out of service or the call control software automatically put them out of service due to a signaling problem (for example, the dial tone was not received from the central office).

— **In Service—**This means the configured ports or IP phones are ready for service.

- **Command:** Allows you to issue commands to the device. The commands are as follows:

— **Restart:** This restarts the voice switch. Active calls are dropped.

This is a forceful way to shed traffic from a voice switch when you are performing a software upgrade.

— **Restart when idle:** This restarts the voice switch once all ports are idle. All idle ports are put out of service and remaining ports are put out of service when they go idle. Once all ports are out of service, the voice switch restarts.

This is a graceful way to shed traffic from a voice switch when you are performing a software upgrade. Note that active calls are completed but no new calls can be made or received until the voice switch restarts.

— **Put in service:** This puts all ports on the voice switch in service. Ports already in service with active calls are not affected.

— **Put out of service:** This places all ports on the voice switch out of service. Active calls are dropped.

This is a forceful way to shed traffic from a voice switch when you need to replace the switch.

— **Put out of service when idle:** All idle ports are put out of service and remaining ports are put out of service when they go idle.

This is a graceful way to shed traffic from a voice switch when you need to replace the switch.

NOTE SoftSwitches only have Restart and Restart When Idle in the command list.

19.1.1.2 Spare Switch Parameters

The columns in the Spare **Switches** section includes the following elements:

- **Spare Switch:** Lists the name of the spare switch.
- **IP Phones:** List number of phones currently registered with the switch and the total number the switch supports.
- **Type:** Lists the switch type.
- **IP Address:** Lists the IP address of the spare switch.
- **MAC Address:** List the Ethernet address of the spare switch.
- **Failover Status:**
- **Current Site:** Lists the site the spare switch is currently supporting.
- Home Site: Lists the site in which the spare switch is registered.
- **Usage:** This is the usage state of the spare switch.

— **Unknown—**The usage state is unknown. This might be true because communication between the server and the switch might have been lost.

— **In Use—**At least one port or IP phone has an active call.

— **Off Hook—**At least one port or IP phone is off-hook, but no ports are in use.

— **Idle—**There are no ports or IP phones off-hook or in-use.

- **Service**: This is the service state of the spare switch.

— **Lost Communication—**The server lost communication with the voice switch. Note that the voice switch may be fully operational but the ShoreTel server cannot see the voice switch due to a networking issue. This also occurs when the voice switch is powered off.

— **Upgrade in Progress—**The voice switch is currently being upgraded with a new software version.

— **Restart Pending—**A Restart when idle command was issued but the restart did not occur because ports are still in use.

— **Firmware Version Mismatch—**The voice switch is running a version of software that does not match the version on the server. The voice switch continues to run call control but does not have access to any voice services on the server. This typically happens on software upgrades after the server was upgraded but before the voice switches were restarted and upgraded. Note that voice switches at the same firmware version also continue to operate together.

— **Firmware Update Available—**This means the server has a new optional version of firmware available for the voice switch. A voice switch in this state continues to run call control as well as access the voice services on the server. This state typically happens when you install a patch on the ShoreTel server. If you want, or need, the patch to propagate to the voice switches, you must restart them.

— **FTP Booted—**This means the ShoreTel voice switch did not boot from FLASH memory but booted from an FTP server, most likely on the ShoreTel server. You can correct this problem by rebooting the voice switch. If this does not correct the problem, please contact ShoreTel's Customer Response Center.

— **Port Out of Service—**This means all ports on the ShoreTel voice switch are out of service.

— **IP Phone(s) Out of Service—**This means one or more IP phones associated with the switch are out of service.

— **Port Out of Service—**This means one or more, but not all, ports or IP phones are out of service on the ShoreTel voice switch. Ports or IP phones typically go out of service because either someone manually put them out of service or the call control software automatically put them out of service due to a signaling problem (for example, the dial tone was not received from the central office).

— **In Service—**This means the configured ports or IP phones are ready for service.

- **Command**: Allows you to issue commands to the device. The commands are as follows:

— **Restart:** This restarts the voice switch. Active calls are dropped.

This is a forceful way to shed traffic from a voice switch when you are performing a software upgrade.

— **Restart when idle:** This restarts the voice switch once all ports are idle. All idle ports are put out of service and remaining ports are put out of service when they go idle. Once all ports are out of service, the voice switch restarts.

This is a graceful way to shed traffic from a voice switch when you are performing a software upgrade. Note that active calls are completed but no new calls can be made or received until the voice switch restarts.

— **Put in service:** This puts all ports on the voice switch in service. Ports already in service with active calls are not affected.

— **Put out of service:** This places all ports on the voice switch out of service. Active calls are dropped.

This is a forceful way to shed traffic from a voice switch when you need to replace the switch.

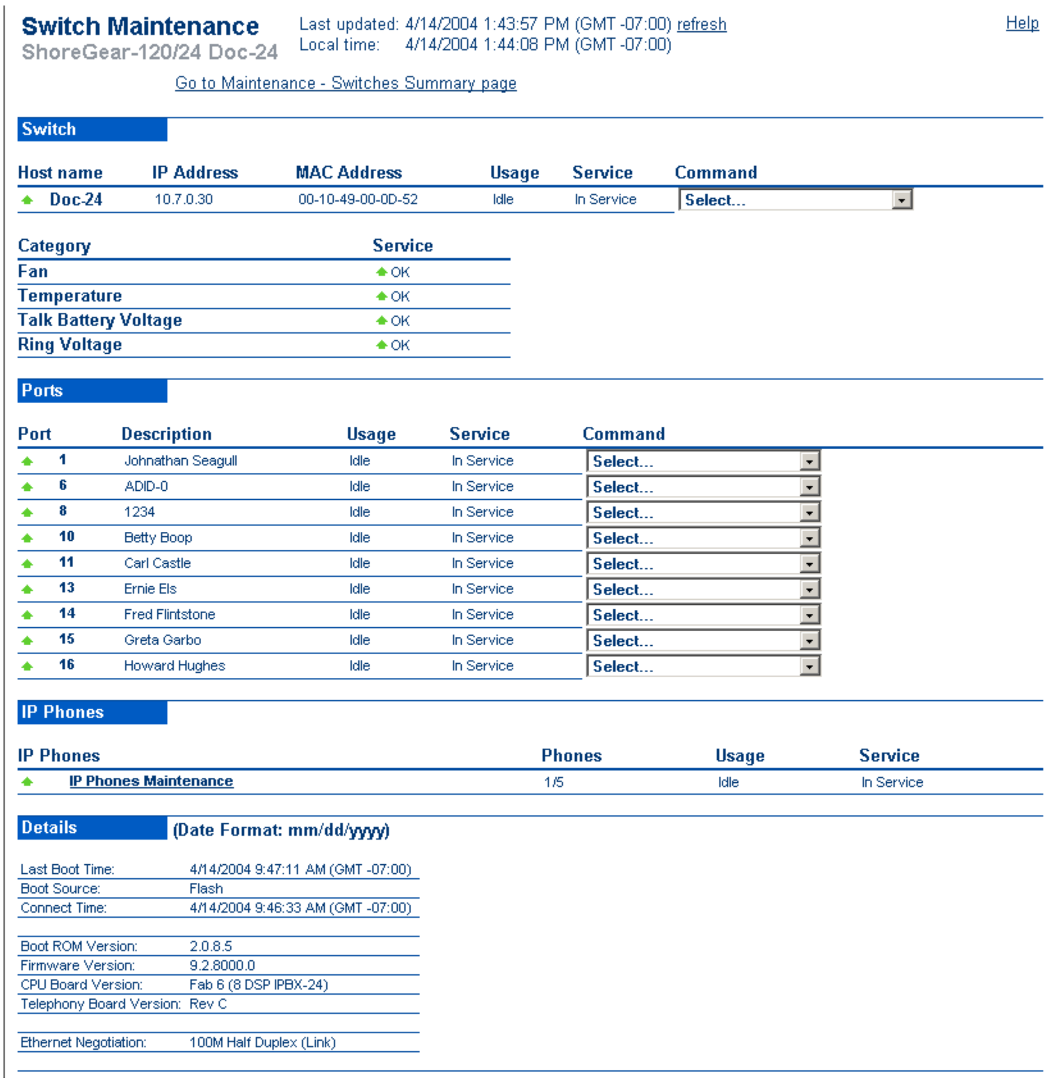
— **Put out of service when idle:** All idle ports are put out of service and remaining ports are put out of service when they go idle.

This is a graceful way to shed traffic from a voice switch when you need to replace the switch.

— **Failback:** Clears the parameters assigned to the system for failover and returns the switch to the spare-switch state at the home site.

19.2 Switch Maintenance

Clicking an individual switch name on the Switches Summary page brings you to the associated Switch Maintenance page. The individual Switch Maintenance pages are very similar but may vary to some degree. Examples of Switch Maintenance pages such as Figure 19-3 are shown on the pages that follow. Each type of switch may show less than all of the details show.



Switch Maintenance
ShoreGear-120/24 Doc-24

Last updated: 4/14/2004 1:43:57 PM (GMT -07:00) refresh
Local time: 4/14/2004 1:44:08 PM (GMT -07:00)

Help

Go to Maintenance - Switches Summary page

Switch

Host name	IP Address	MAC Address	Usage	Service	Command
Doc-24	10.7.0.30	00-10-49-00-0D-52	Idle	In Service	Select...

Category	Service
Fan	OK
Temperature	OK
Talk Battery Voltage	OK
Ring Voltage	OK

Ports

Port	Description	Usage	Service	Command
1	Johnathan Seagull	Idle	In Service	Select...
6	ADID-0	Idle	In Service	Select...
8	1234	Idle	In Service	Select...
10	Betty Boop	Idle	In Service	Select...
11	Carl Castle	Idle	In Service	Select...
13	Ernie Els	Idle	In Service	Select...
14	Fred Flintstone	Idle	In Service	Select...
15	Greta Garbo	Idle	In Service	Select...
16	Howard Hughes	Idle	In Service	Select...

IP Phones

IP Phones	Phones	Usage	Service
IP Phones Maintenance	1/5	Idle	In Service

Details (Date Format: mm/dd/yyyy)

Last Boot Time:	4/14/2004 9:47:11 AM (GMT -07:00)
Boot Source:	Flash
Connect Time:	4/14/2004 9:46:33 AM (GMT -07:00)
Boot ROM Version:	2.0.8.5
Firmware Version:	9.2.8000.0
CPU Board Version:	Fab 6 (8 DSP IPBX-24)
Telephony Board Version:	Rev C
Ethernet Negotiation:	100M Half Duplex (Link)

Figure 19-3 Switch Maintenance Page for a ShoreGear 120/24 Switch

19.2.1 Switch Section

The top of the Switch Maintenance page repeats some of the information on the Switches Summary page. The Switch Maintenance page also expose the individual ports as well as more detailed information on the switch operation.

The following is additional state information for ShoreTel-120/24, ShoreTel-60/12, ShoreTel-40/8, ShoreTel-T1, and ShoreTel-E1:

- **Fan**: Status of the fan, with the following states:
  - — OK
  - — Slow
  - — Failed
  - — Unknown

  Switches: 24, 12, 8, T1, E1

- **Temperature:** Status of the temperature, with the following states:
  - — OK
  - — Yellow Alarm
  - — Red Alarm
  - — Unknown

  Switches: 24, 12, 8, T1, E1

- **Talk Battery Voltage:** Status of the Talk Battery Voltage, with the following states:
  - — OK
  - — Failed
  - — Unknown

  Switches: 24, 12, 8

- **Ring Voltage:** Status of the Ring Voltage, with the following states:
  - — OK
  - — Failed
  - — Unknown

  Switches: 24, 12, 8

19.2.2 Link Section

The ShoreTel-T1 and ShoreTel-E1 uses T1, T1 PRI, or E1 PRI signaling and additional information is presented on the status of the link and on the status of the PRI D-Channel.

Figure 19-4 displays the Switch Maintenance page for a ShoreTel T1 switch.

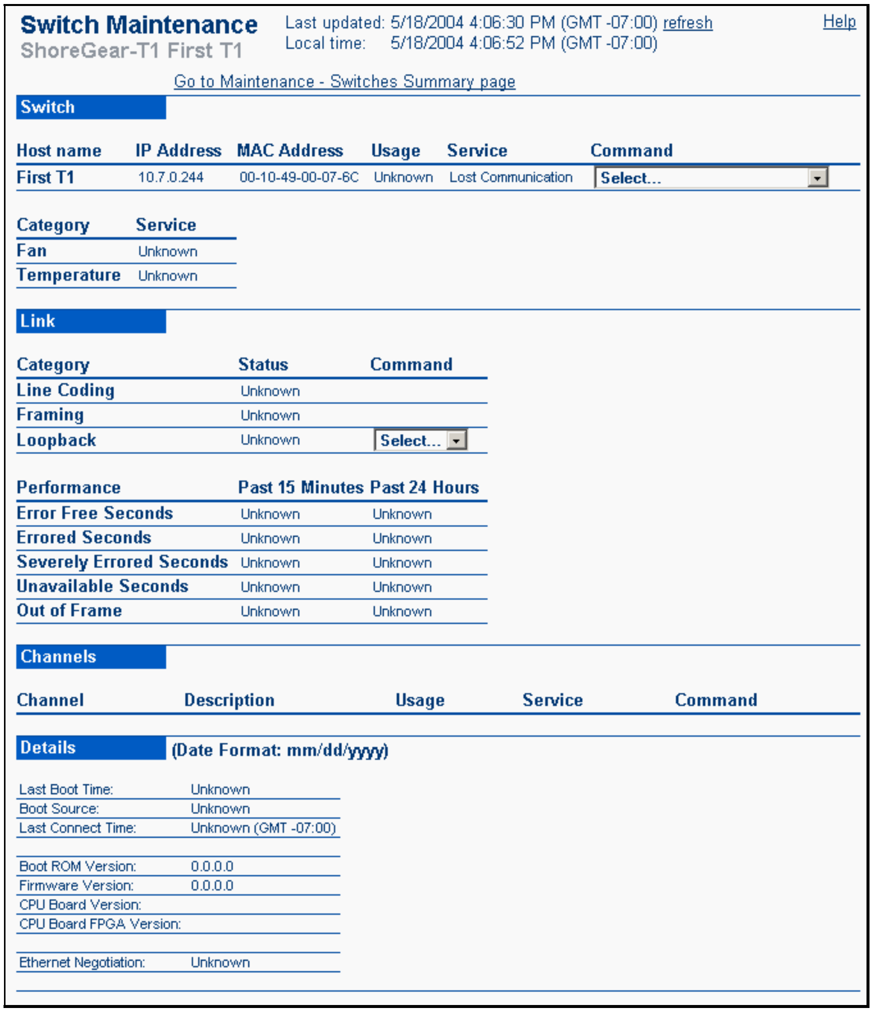


Figure 19-4 Link Section of the Switch Maintenance Page (SG-T1)

Figure 19-5 displays the Switch Maintenance page for a ShoreTel 220E1 switch.

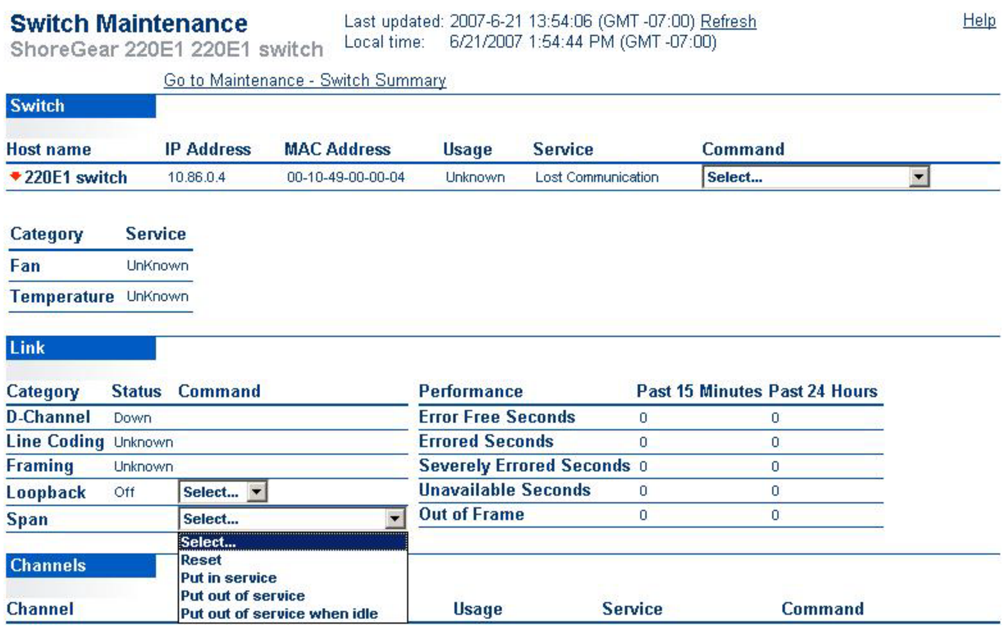


Figure 19-5 Link Section of the Switch Maintenance Page (SG-220E1)

The following is the link state information for the SG-T1/E1 and SG-220T1/220E1:

- **D-Channel (PRI only)**
  - **States:** In Service, Out of Service, Unknown
  - **Commands:** Reset, Put in service, Put out of service, Put out of service when idle
- **Line Coding**
  - **States:** OK, Bipolar Violations, Loss of Signal, Unknown
- **Framing**
  - **States:** OK, Yellow Alarm, Bit Error, Out of Frame, Unknown
- **Loopback**
  - **States:** Off, On, Unknown
  - **Commands:** Turn On, Turn Off
- **Span**
  - **Commands:** Reset, Put in service, Put out of service, Put out of service when idle.

The following is the link performance information for the SG-T1/E1 and SG-220T1/220E1:

- **Error Free Seconds:** The number of error-free seconds that occurred in the last 15 minutes and 24 hours.
- **Errored Seconds:** The number of errored seconds that occurred in the last 15 minutes and 24 hours.
- **Severely Errored Seconds:** The number of severely errored seconds that occurred in the last 15 minutes and 24 hours.
- **Unavailable Seconds:** The number of seconds the server was not available.

- **Bit Error Rate:** Not used.
- **Out of Frame:** The number of times the link has been out of frame in the past 15 minutes and 24 hours.

19.2.3 Ports Section

Each Switch Maintenance page has a Ports section, where the configured ports are shown. The Ports section shows the port number, port description (first and last name or trunk name), usage state, service state, and available commands. Refer to Figure 19-6.

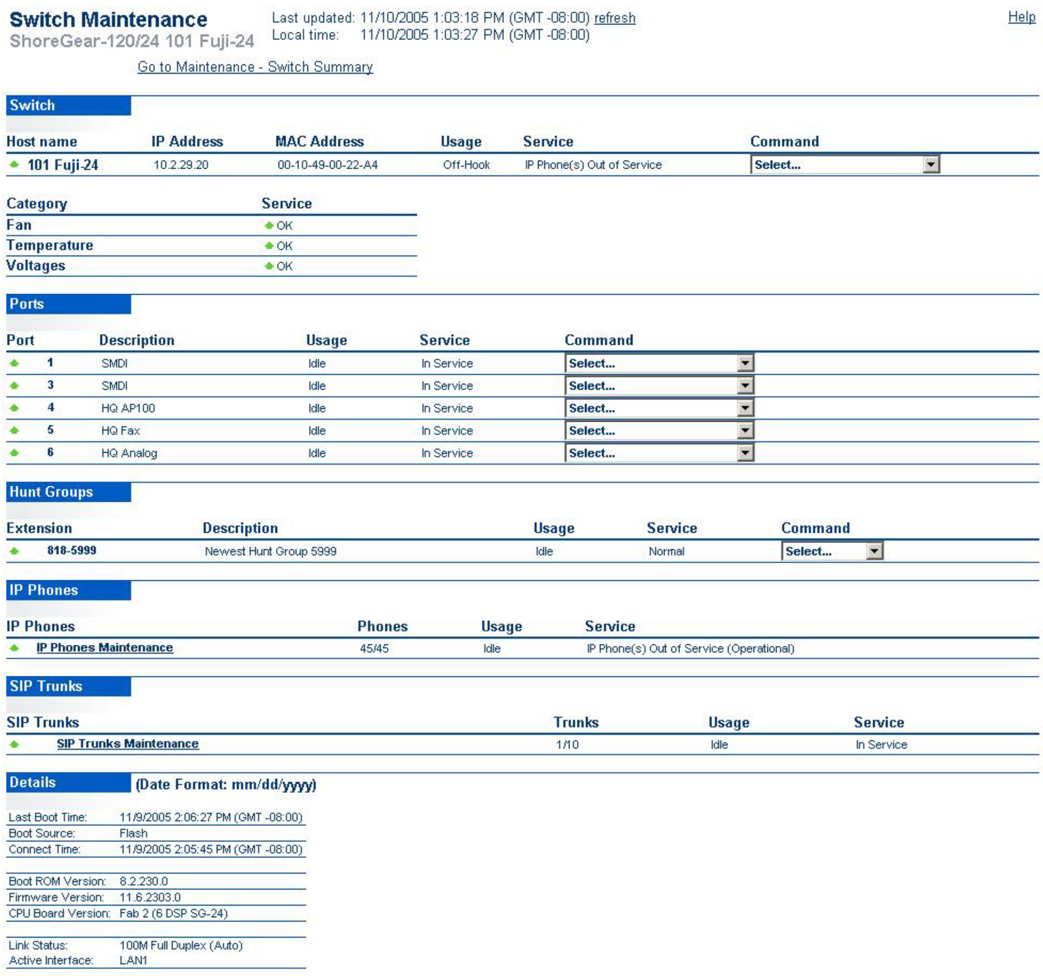


Figure 19-6 Ports, Hunt Groups, and Details Sections of Switch Maintenance Page

The following are the port **Usage** states:

- **Unknown:** The state is unknown, likely because communication between the server and the switch has been lost.
- **In Use:** The port has at least one active call.
- **Off Hook:** The port is off-hook and has no active calls.

The following are the port **Service** states:

- **In Service (green arrow up):** The port is in service.

- **Out of Service (red arrow down):** The port is out of service.

  On telephone ports, outbound calls cannot be made (no dial tone) and inbound calls will not be offered.

  On trunk ports, outbound calls will not seize the trunk, and inbound calls are not answered. On loop start trunks, the trunk is seized to emulate a busy condition to the central office.

- **Out of Service (operational) (red arrow down):** The port is out of service due to a manual “put out of service” command.

  Applies only to trunks. The trunk failed to seize. The switch automatically attempts to seize the trunk on a periodic basis. When successful, the trunk is automatically put back in service.

The following are the port Commands:

- **Reset:** Resets the port (puts the port out of service then automatically back in service). Active calls are dropped.
- **Put in service:** Puts the port back in service.
- **Put out of service:** Puts the port out of service. Active calls are dropped.
- **Put out of service when idle:** Puts an idle port out of service and puts an in-use port out of service when it goes idle.

19.2.4 Hunt Groups Section

Double-clicking on a Hunt Group on the Switch Maintenance page shows the Hunt Groups edit page as shown in Figure 19-7. You can use the Command drop-down to busy out a hunt group or return it to normal service from this page.

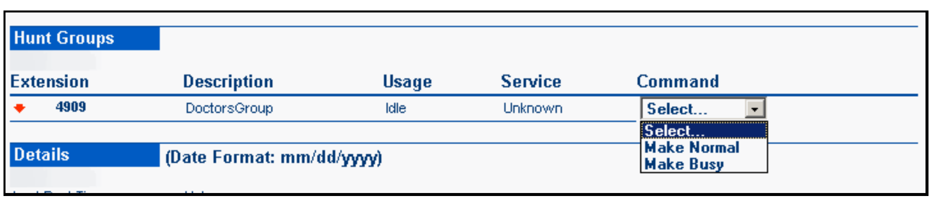


Figure 19-7 Changing State of a Hunt Group

This section shows the number of Hunt Groups configured on the Switch, The information provided in the Hunt Groups section includes:

- **Extension:** The extension number of the hunt group.
- **Description:** The text name of the hunt group.
- **Usage:** The current usage state of the hunt group: Idle or Normal
- **Service:** Current service state of the hunt group: In Service, Out of Service, Unknown
- **Command:** Choose between Normal or Make Busy. Selecting Make Busy idles out the Hunt Group.

19.2.5 IP Phone Maintenance

The ShoreTel-120/24 Switch Maintenance page shown in Figure 19-8 has an additional section for IP phones. It shows how many IP phones are connected through the switch and the capacity of the switch. ShoreTel-120/24 switches can support as many as 120 IP phones, the ShoreTel 60/12 can support up to 60 IP phones, and the ShoreTel-40/8 can support up to 40 IP phones

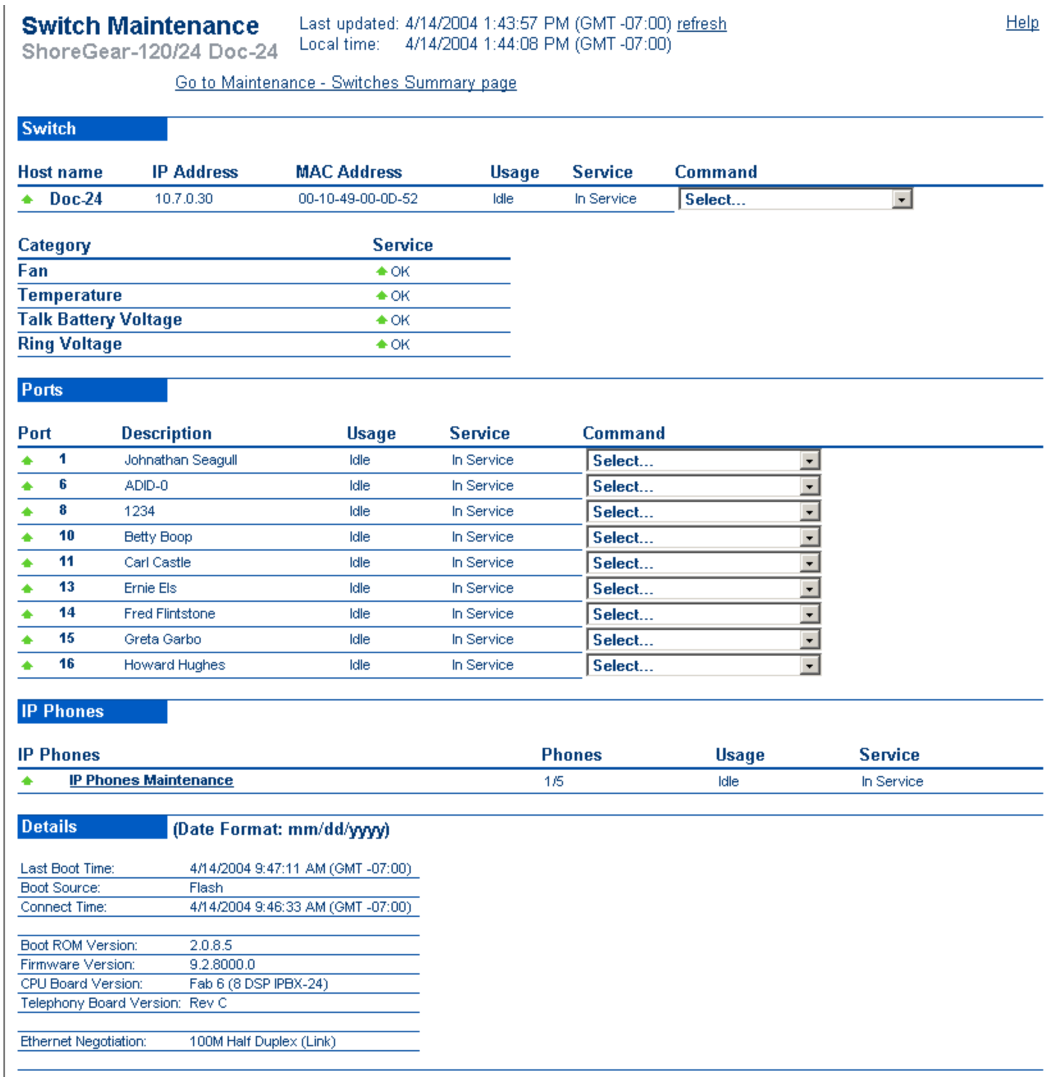


Figure 19-8 IP Phones Section of Switch Maintenance Page

- **Phones:** Lists the number of IP phones on the switch.
- **Usage:** The current usage state of the IP phones: Idle or Normal
- **Service:** Current service state of the IP phones: In Service, Out of Service, Unknown

You can perform several functions on the IP phones connected to the switch by double-clicking the IP Phones Maintenance link. Figure 19-9 shows the IP Phones Maintenance page.

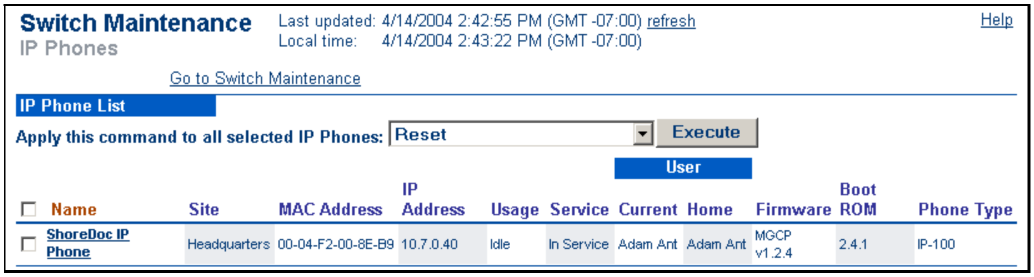


Figure 19-9 IP Phones Maintenance Page

You can view the service history of a phone by clicking on its name from the IP Phone Maintenance page.

19.2.6 SIP Trunks Section

The Switch Maintenance page shown in Figure 19-10 has an additional section for SIP trunks. This section shows how many SIP trunks are connected through the switch and the capacity of the switch.

Figure 19-10 SIP Trunks Maintenance Page

- **Trunks:** Lists the number of SIP trunks on the switch.
- **Usage:** The current usage state of the trunk group: Idle or Normal
- **Service:** Current service state of the trunk group: In Service, Out of Service, Unknown

19.2.7 Details Section

The following additional information is provided in the ***Details*** section:

- **Last Boot Time:** The last time the switch booted.
- **Boot Source:** Flash, FTP boot, or unknown boot source.
- **Connect Time:** The last time the server re-established a connection with the switch.
- **Boot ROM Version:** The boot ROM version number. Recommended 2.0.x.x or higher.
- **Firmware Version:** The firmware version number the voice switch is running.
- **CPU Board Version:** The version number of the switch's CPU board.
- **Telephony Board Version:** The version number of the telephony board.

- **Ethernet Negotiation:** The rate at which the voice switch negotiated the Ethernet interface. Ethernet states include:
  - — 10: Half-Duplex
  - — 10: Full-Duplex
  - — 100: Half
  - — 100: Full

19.2.8 ShoreTel-T1 PRI and ShoreTel-E1 PRI Maintenance

When you click the name of a ShoreTel-T1 switch configured for PRI of the ShoreTel-E1 on the Switches Summary page, the Switch Maintenance page shown in Figure 19-11 appears.

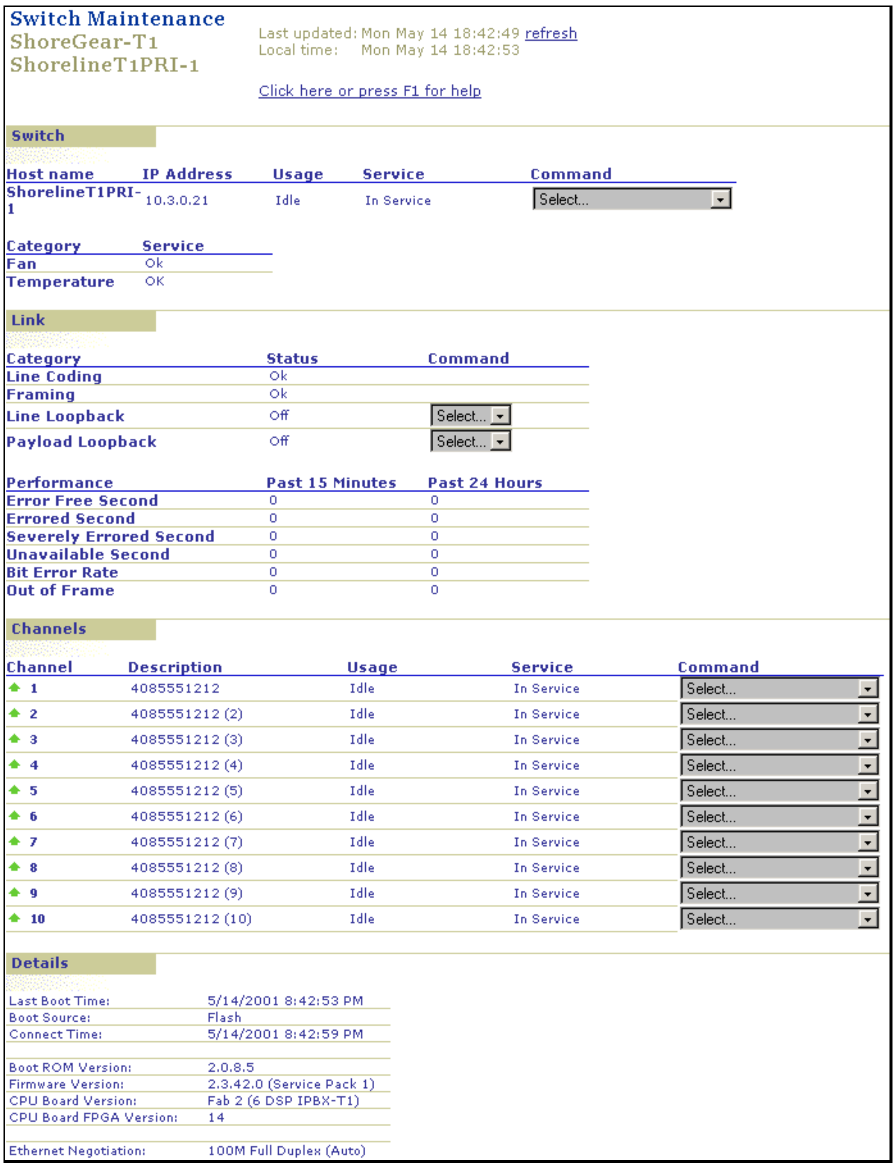


Figure 19-11 ShoreTel-T1 PRI or ShoreTel-E1 PRI Switch Maintenance Page

19.2.9 Quick Look Server Maintenance—Main Server

Click a server from the list on the Quick Look Server Maintenance page. The main server is shown in Figure 19-12. It is divided into three sections: Status, Services, and Events.

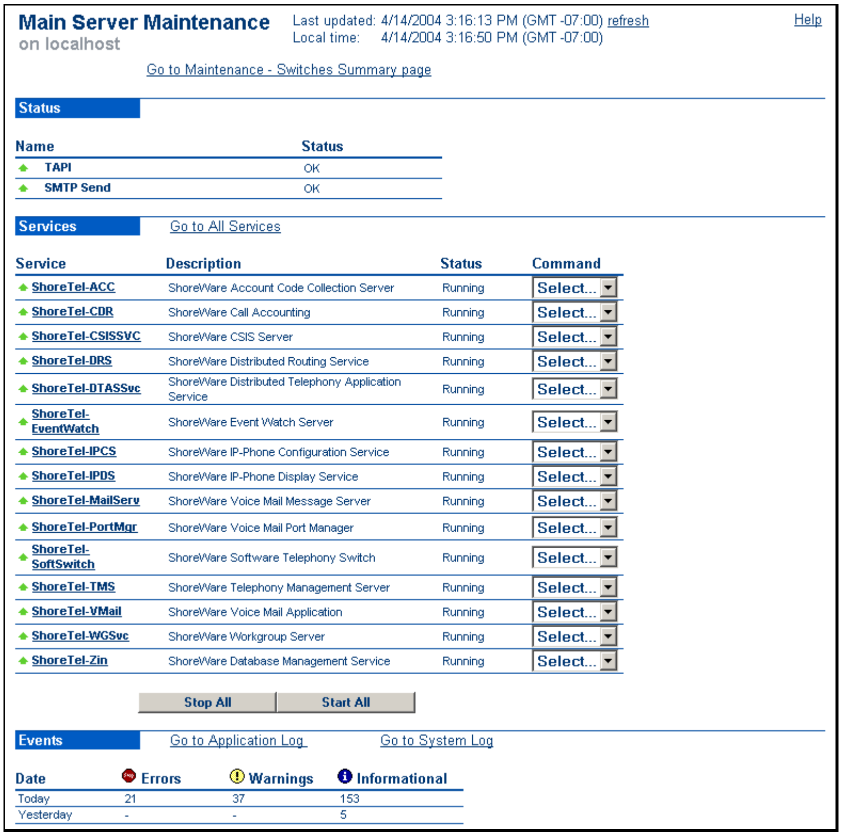


Figure 19-12 Quick Look Main Server Maintenance

19.2.9.1 Quick Look Server Status Area

The information displayed in the Status areas is as follows:

- TAPI States:
  - **OK**—The TSP on the main server knows that the applications on the server have logged into TAPI
  - **Lost TAPI**—The TSP on the main server knows that the applications on the server are not logged into TAPI.
- SMTP Send States:
  - **OK**—The SMTP messages for the server are leaving.
  - **Failed**—The SMTP outbound queue on the server is not sending e-mail messages. The SMTP transport is used to send voice messages between voice mail servers.

19.2.9.2 Quick Look Server Services Area

The Services area provides details about the ShoreTel services areas shown in Table 19-1.

Table 19-1 ShoreTel Service Areas

Name	Description	Details
ShoreTel-ACC	ShoreTel Account Codes Collection Server	This service collects call information by account code and reports call activity in the call detail reports.
ShoreTel-CDR	ShoreTel Call Accounting	This service records call accounting information, call queueing data, and media stream data.
ShoreTel-CSISSVC	ShoreTel CSIS Server	This service manages communications between the server and ShoreTel Communicator.
ShoreTel-DTASSvc	ShoreTel Distributed Telephony Application Service	This service provides connectivity between applications and instances of distributed TMS.
ShoreTel-DRS	ShoreTel Distributed Routing Service	This service allows the ShoreTel system to scale beyond 100 switches.
ShoreTel-EventWatch	ShoreTel EventWatch Server	This service monitors the event log and delivers e-mail notifications on certain events.
ShoreTel-IPCS	ShoreTel IP Phone Configuration Service	This service automatically configures IP phones as they are inserted into the system.
ShoreTel-IPDS	ShoreTel IP Phone Display Service	This service manages the IP phone display.
ShoreTel-MailServ	ShoreTel Voice Mail Message Server	This service is part of the ShoreTel Voice Mail system.
ShoreTel-PortMgr	ShoreTel Voice Mail Port Manager	This service is part of the ShoreTel Voice Mail system.
ShoreTel-SoftSwitch	ShoreTel Software Telephony Switch	This is the ShoreTel call control software running the server generally used to host virtual users.
ShoreTel-TMS	ShoreTel Telephony Management Server	This service provides the telephony platform for ShoreTel applications and services.
ShoreTel-Vmail	ShoreTel Voice Mail Application	This service is part of the ShoreTel Voice Mail system.
ShoreTel-WGSvc	ShoreTel Workgroup Server	This service manages workgroups and queues.
ShoreTel-Zin	ShoreTel Database Management Service	This service manages and updates the ShoreTel database.

Clicking the ShoreTel service name takes you to a page used for starting or stopping that service, as shown in Figure 19-13.

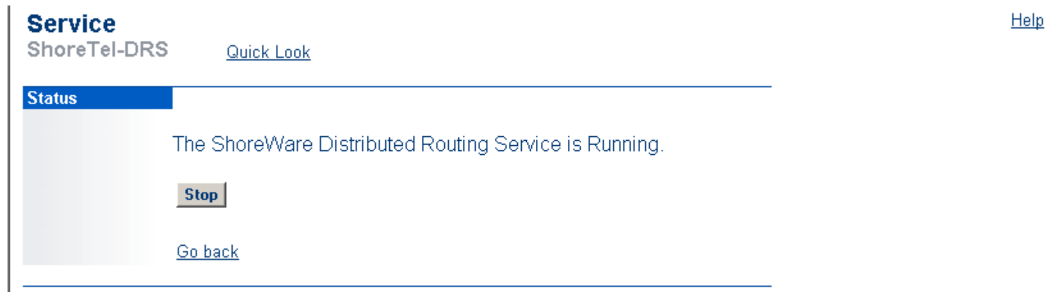


Figure 19-13 Service Maintenance Page

19.2.9.3 Quick Look Server Events

The Quick Look Events area provides a quick summary of event activity and lists all events that have been logged during the last two days—Today and Yesterday. Clicking the Application Log icon takes you to the Application Event Log maintenance page. See the "Event Log" on page 558 for more information.

The information displayed in the Quick Look Events area includes:

- **Date:** The day the event was reported: Today or Yesterday.
- **Errors:** The number of errors that were reported. Errors require your immediate attention.
- **Warnings:** The number of warning messages that were reported. Warnings alert you to potential errors.
- **Informational:** The number of informational messages that were reported. Informational messages provide the status of a service, switch, or port.

19.2.10 Quick Look Server Maintenance—Distributed Servers

The Quick Look Distributed Server Maintenance page is shown in Figure 19-14. This page is hosted on a web site on the distributed server. The information provided is a subset of the main server maintenance page, since not all ShoreTel services run on the distributed server.

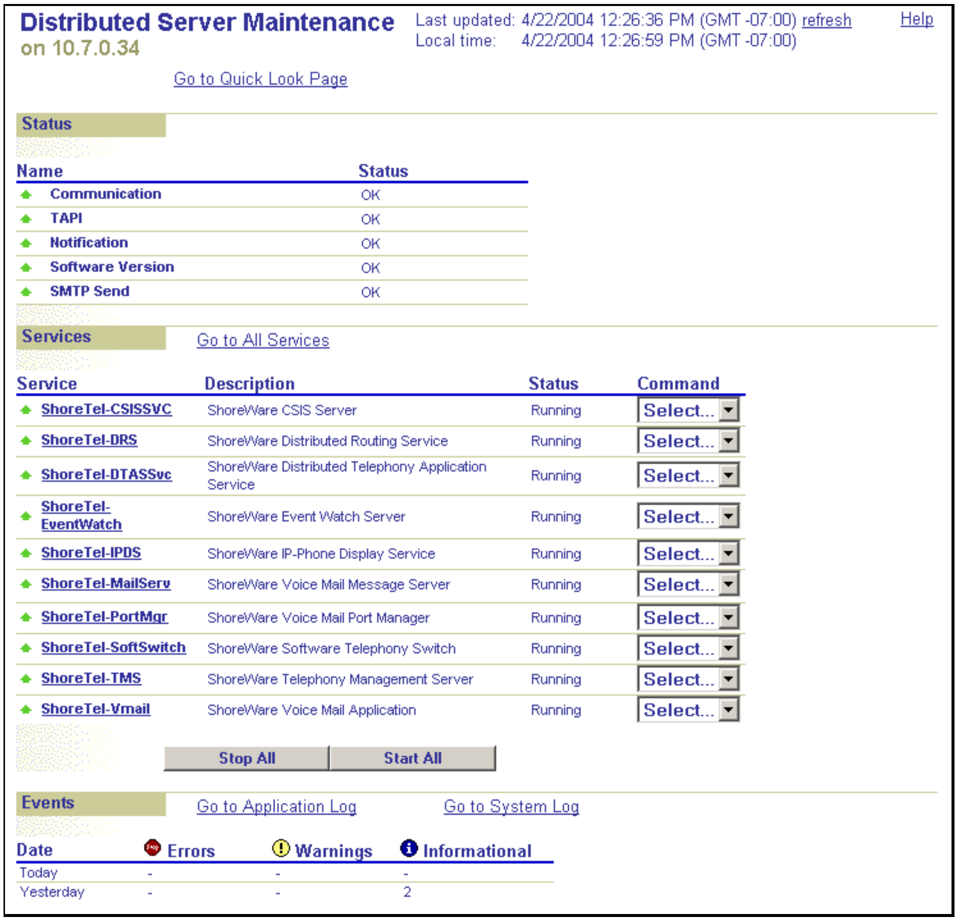
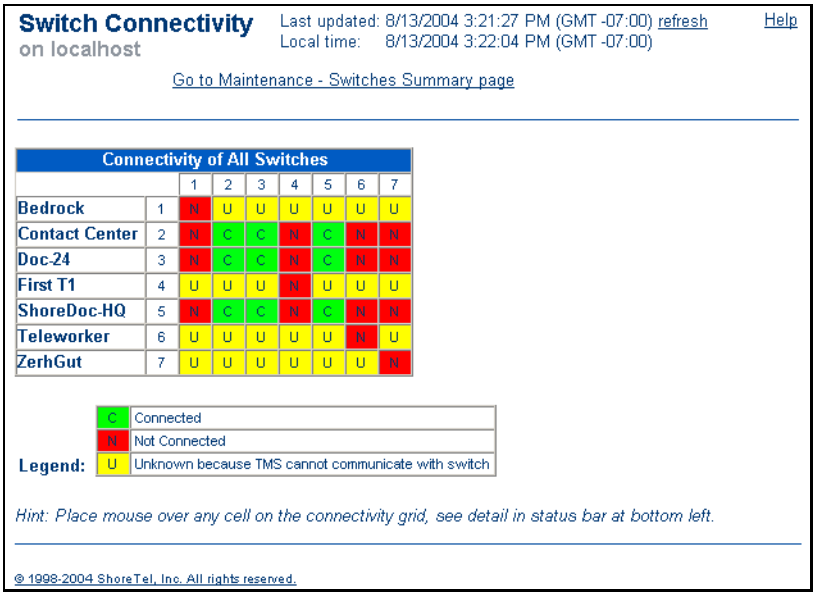


Figure 19-14 Quick Look Server Maintenance—Distributed Server

19.3 Switch Connectivity

The **Switch Connectivity** page (Figure 19-15) lists all ShoreTel voice switches when Distributed Routing Service (DRS) is disabled. When DRS is enabled, the switch connectivity table is organized by site.



		1	2	3	4	5	6	7
Bedrock	1	N	U	U	U	U	U	U
Contact Center	2	N	C	C	N	C	N	N
Doc-24	3	N	C	C	N	C	N	N
First T1	4	U	U	U	N	U	U	U
ShoreDoc-HQ	5	N	C	C	N	C	N	N
Teleworker	6	U	U	U	U	U	N	U
ZerhGut	7	U	U	U	U	U	U	N

Figure 19-15 Switch Connectivity Page

The Switch Connectivity page includes the following:

- **Connectivity Grid:** The following indicators provide information about the listed switch:
  - — Green (C) indicates that the switch is connected and communicating with other switches in the system.
  - — Yellow (U) indicates that the switch connectivity is unknown because it cannot communicate with TMS.
  - — Red (N) indicates that the switch has lost communications with the server.

For more detailed information about a switch, place the mouse cursor over the connectivity cell in question and status information is displayed in the status bar at the bottom of your browser.

19.4 Conference Ports

The Conference Ports page as shown in Figure 19-16shows the number of ports currently being used for conferencing and the number of free ports allocated for conferencing for each site in the system. The number of Make Me Conference calls active at each site is also shown. As with the Quick Look page, clicking on an entry in the Site column displays the Maintenance Switches Summary page described earlier in this chapter (refer to Figure 19-2 on page 539).

Conference Ports Last updated: 8/13/2004 1:16:51 PM (GMT -07:00) refresh Help
Local time: 8/13/2004 1:16:57 PM (GMT -07:00)
Go to Quick Look page

	In-Use		Free	
Site	Active Calls	Ports	Ports	Percent
Headquarters	0	0	4	100%
El Dorado	0	0	0	0%
Remote-Site	0	0	0	0%
Bonn	0	0	0	0%
Total	0	0	4	100%

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Figure 19-16 Conference Ports Page

There are two main section displayed on the Conference Ports page. The first is data about in-use ports and the second is data about free ports.

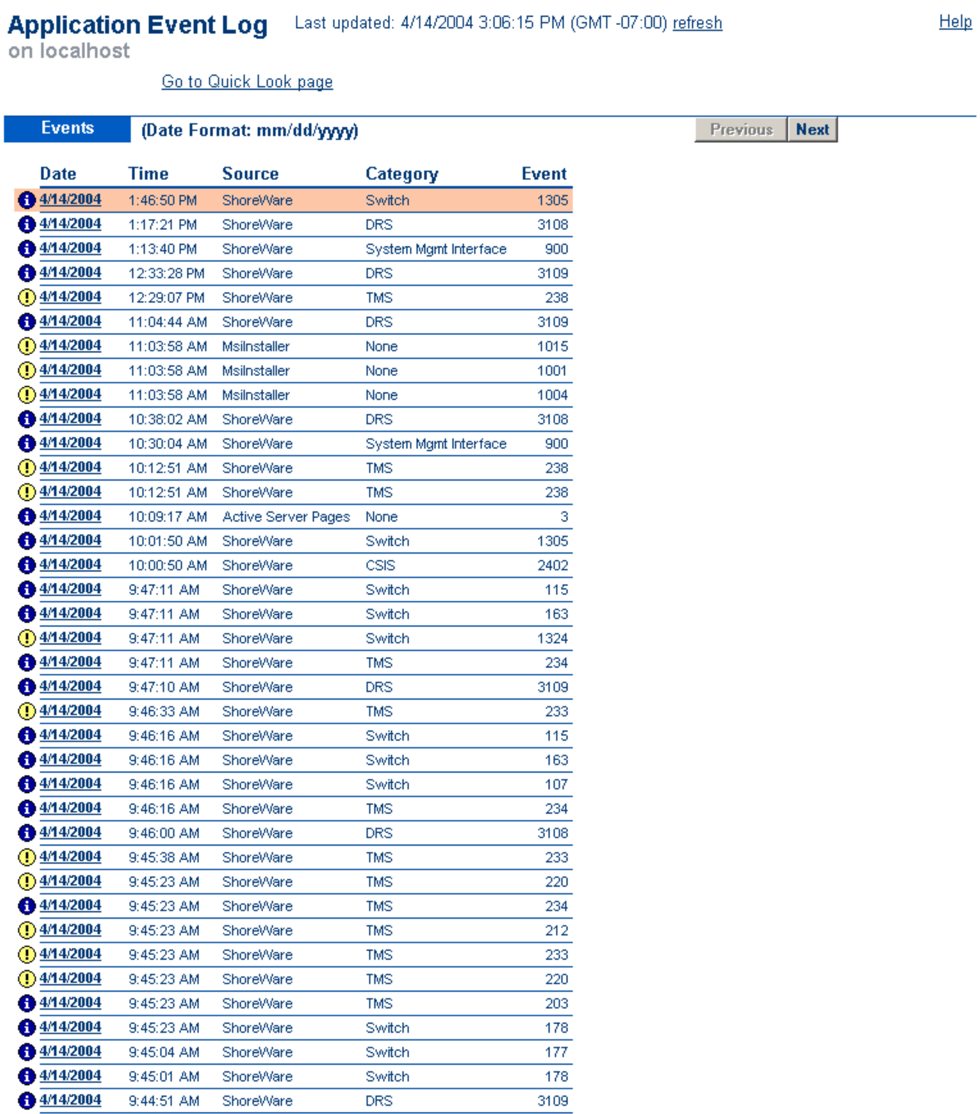
- **In-Use**
  - — **Active Calls**: Lists the number of calls active on the ports at a site
  - — **Ports**: The number of ports that are in use at each site.
- **Free**
  - — **Ports**: The number of ports that are free at each site.
  - — **Percent**: The percentage of total ports at a site that are currently free.

19.5 Event Log

All events are reported to and viewed from the System Event Log (Figure 19-17) and Application Event Log (Figure 19-18) maintenance pages. To refresh these pages, click the **refresh** link. Clicking the Quick Look link takes you to the Quick Look maintenance page, described in "Quick Look" on page 537.

Date	Time	Source	Category	Event
5/18/2004	3:17:30 PM	Schannel	None	36871
5/18/2004	3:17:30 PM	Schannel	None	36871
5/18/2004	3:16:12 PM	ShoreWare-Media	Media Driver	2100
5/18/2004	3:15:56 PM	EventLog	None	6005
5/18/2004	3:15:56 PM	EventLog	None	6009
5/18/2004	3:14:41 PM	EventLog	None	6006
5/18/2004	12:00:52 PM	Schannel	None	36871
5/18/2004	12:00:52 PM	Schannel	None	36871
5/18/2004	11:30:42 AM	ShoreWare-Media	Media Driver	2100
5/18/2004	11:30:33 AM	DCOM	None	10002
5/18/2004	11:30:28 AM	EventLog	None	6005
5/18/2004	11:30:28 AM	EventLog	None	6009
5/18/2004	11:29:05 AM	EventLog	None	6006
5/18/2004	11:28:36 AM	DCOM	None	10002
5/18/2004	11:28:05 AM	DCOM	None	10002
5/18/2004	11:27:35 AM	DCOM	None	10002
5/18/2004	11:27:05 AM	DCOM	None	10002
5/18/2004	11:26:35 AM	DCOM	None	10002
5/18/2004	11:26:05 AM	DCOM	None	10002
5/18/2004	11:25:35 AM	DCOM	None	10002
5/18/2004	11:25:05 AM	DCOM	None	10002
5/18/2004	11:24:35 AM	DCOM	None	10002
5/18/2004	11:24:05 AM	DCOM	None	10002
5/18/2004	11:23:32 AM	DCOM	None	10002
5/18/2004	11:23:02 AM	DCOM	None	10002
5/18/2004	11:22:32 AM	DCOM	None	10002
5/18/2004	11:22:02 AM	DCOM	None	10002
5/18/2004	11:21:32 AM	DCOM	None	10002
5/18/2004	11:21:02 AM	DCOM	None	10002
5/14/2004	1:56:28 AM	Schannel	None	36871
5/14/2004	1:56:28 AM	Schannel	None	36871
5/13/2004	8:10:13 PM	ShoreWare-Media	Media Driver	2100
5/13/2004	8:09:57 PM	EventLog	None	6005
5/13/2004	8:09:57 PM	EventLog	None	6009
5/13/2004	8:08:43 PM	EventLog	None	6006
5/13/2004	8:08:32 PM	MRxSmb	None	3034
5/13/2004	8:08:32 PM	MRxSmb	None	3034
5/13/2004	8:08:32 PM	BROWSER	None	8021

Figure 19-17 System Event Log Page



Application Event Log Last updated: 4/14/2004 3:06:15 PM (GMT -07:00) refresh Help
on localhost

Go to Quick Look page

Events (Date Format: mm/dd/yyyy) Previous Next

Date	Time	Source	Category	Event
4/14/2004	1:46:50 PM	ShoreWare	Switch	1305
4/14/2004	1:17:21 PM	ShoreWare	DRS	3108
4/14/2004	1:13:40 PM	ShoreWare	System Mgmt Interface	900
4/14/2004	12:33:28 PM	ShoreWare	DRS	3109
4/14/2004	12:29:07 PM	ShoreWare	TMS	238
4/14/2004	11:04:44 AM	ShoreWare	DRS	3109
4/14/2004	11:03:58 AM	MsiInstaller	None	1015
4/14/2004	11:03:58 AM	MsiInstaller	None	1001
4/14/2004	11:03:58 AM	MsiInstaller	None	1004
4/14/2004	10:38:02 AM	ShoreWare	DRS	3108
4/14/2004	10:30:04 AM	ShoreWare	System Mgmt Interface	900
4/14/2004	10:12:51 AM	ShoreWare	TMS	238
4/14/2004	10:12:51 AM	ShoreWare	TMS	238
4/14/2004	10:09:17 AM	Active Server Pages	None	3
4/14/2004	10:01:50 AM	ShoreWare	Switch	1305
4/14/2004	10:00:50 AM	ShoreWare	CSIS	2402
4/14/2004	9:47:11 AM	ShoreWare	Switch	115
4/14/2004	9:47:11 AM	ShoreWare	Switch	163
4/14/2004	9:47:11 AM	ShoreWare	Switch	1324
4/14/2004	9:47:11 AM	ShoreWare	TMS	234
4/14/2004	9:47:10 AM	ShoreWare	DRS	3109
4/14/2004	9:46:33 AM	ShoreWare	TMS	233
4/14/2004	9:46:16 AM	ShoreWare	Switch	115
4/14/2004	9:46:16 AM	ShoreWare	Switch	163
4/14/2004	9:46:16 AM	ShoreWare	Switch	107
4/14/2004	9:46:16 AM	ShoreWare	TMS	234
4/14/2004	9:46:00 AM	ShoreWare	DRS	3108
4/14/2004	9:45:38 AM	ShoreWare	TMS	233
4/14/2004	9:45:23 AM	ShoreWare	TMS	220
4/14/2004	9:45:23 AM	ShoreWare	TMS	234
4/14/2004	9:45:23 AM	ShoreWare	TMS	212
4/14/2004	9:45:23 AM	ShoreWare	TMS	233
4/14/2004	9:45:23 AM	ShoreWare	TMS	220
4/14/2004	9:45:23 AM	ShoreWare	TMS	203
4/14/2004	9:45:23 AM	ShoreWare	Switch	178
4/14/2004	9:45:04 AM	ShoreWare	Switch	177
4/14/2004	9:45:01 AM	ShoreWare	Switch	178
4/14/2004	9:44:51 AM	ShoreWare	DRS	3109

Figure 19-18 Application Event Log Page

ShoreTel system events can also be viewed with the Windows Event Viewer.

Event categories for ShoreTel sources are reported to the applications log, in the following categories:

- Event Watch
- Java Client
- Java Server
- Notification
- Port Mapper
- Software Telephony Switch
- Switch
- System Management Database

- System Management Interface
- Voice Mail Application
- Voice Mail Message Server
- Voice Mail Port Mapper
- Workgroup Server

19.5.1 Viewing Event Reports

To view an event report, click an entry in the Date column on the Services or Application Event Log page. The associated Event Info page is invoked as shown in Figure 19-19.

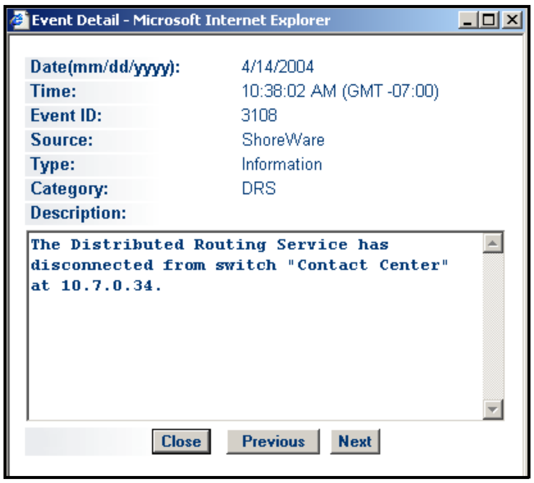


Figure 19-19 Event Info Page

The Event Info page provides the following information, from which you can ascertain the severity of the event:

- Source reporting the event
- Category of the event
- Event number
- Event description
- Date and time
- Event type (error, warning, or information)

Clicking the **Go back** link takes you back to the top-level event log (**System** or **Application**) page.

19.6 Services

The Services maintenance page, shown in Figure 19-20, lists all the services that reside on the server. It includes the name of the service, its description, and its operational status (running, not running, or paused).

Services on localhost

Last updated: 4/14/2004 3:10:35 PM (GMT -07:00) refresh
Local time: 4/14/2004 3:10:44 PM (GMT -07:00)

Go to Quick Look page

Help

Services

Name	Description	Status
Alerter	Alerter	Running
AppMgmt	Application Management	Not Running
BITS	Background Intelligent Transfer Service	Running
Browser	Computer Browser	Running
cisvc	Indexing Service	Not Running
ClipSrv	ClipBook	Not Running
Dfs	Distributed File System	Running
Dhcp	DHCP Client	Running
dmadmin	Logical Disk Manager Administrative Service	Not Running
dmserver	Logical Disk Manager	Running
Dnscache	DNS Client	Running
Eventlog	Event Log	Running
EventSystem	COM+ Event System	Running
Fax	Fax Service	Not Running
IISADMIN	IIS Admin Service	Running
IsmServ	Intersite Messaging	Not Running
kdc	Kerberos Key Distribution Center	Not Running
lanmanserver	Server	Running
lanmanworkstation	Workstation	Running
LicenseService	License Logging Service	Running
LmHosts	TCP/IP NetBIOS Helper Service	Running
McAfeeFramework	McAfee Framework Service	Running
McShield	Network Associates McShield	Running
mcsvc	Monitor Control Service	Not Running
McTaskManager	Network Associates Task Manager	Running
Messenger	Messenger	Running
mnmsrvc	NetMeeting Remote Desktop Sharing	Not Running
MSDTC	Distributed Transaction Coordinator	Running
MSFTPSVC	FTP Publishing Service	Running
MSIServer	Windows Installer	Not Running
NetDDE	Network DDE	Not Running
NetDDEdsdm	Network DDE DSDM	Not Running
Netlogon	Net Logon	Not Running
Netman	Network Connections	Running
NtFrs	File Replication	Not Running
NtLmSsp	NT LM Security Support Provider	Not Running
NtmsSvc	Removable Storage	Running

Figure 19-20 Services Page

To refresh the page, click the refresh link. Clicking the Quick Look link takes you to the Quick Look maintenance page, described in "Quick Look" on page 537.

Green, upward-pointing arrows indicate that the service or application is running. Red, downward-pointing arrows indicate that the service or application is not running or paused.

19.6.1 Service Maintenance Page

To start a service that is not running or paused, click the service name to invoke the Service Maintenance page, shown in Figure 19-21.

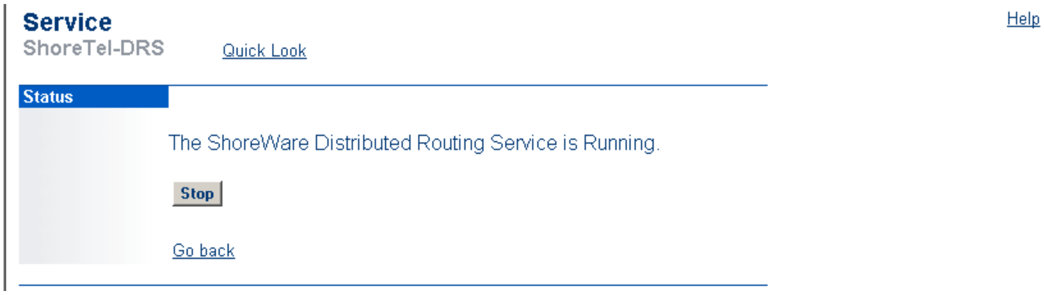


Figure 19-21 Service Maintenance Page

From this page you can start or stop the service by clicking the Start or Stop button. Clicking the Quick Look link takes you back to the Quick Look maintenance page, described in "Quick Look" on page 537. Clicking the Go back link takes you back to the Services page.

19.7 Event Filters

Event filters specify the criteria for which e-mail notifications are sent after an event has been reported. The Event Filters list page, shown in Figure 19-22, displays a list of event filters that you create. You invoke this page by clicking Event Filters from under the Maintenance link in the navigation frame.

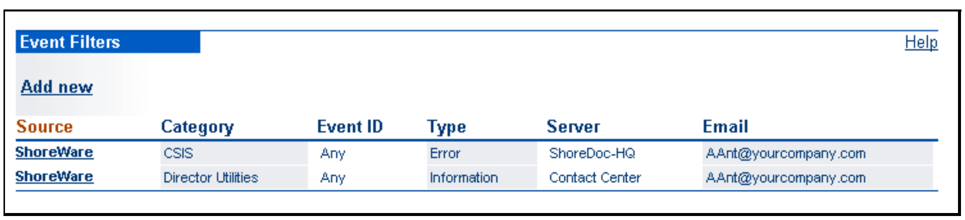


Figure 19-22 Event Filters List Page

The Event Filters list page provides the following information:

- **Source:** The name of the event source.
- **Category:** The category for the source.
- **Event ID:** The event number that is assigned to the filter.
- **Type:** The event type that is assigned to the filter.
- **Server:** The server that the event filter runs on.
- **Email:** The e-mail address that the filter reports to.

To view the parameters of an existing event filter, click its name in the *Source* column.

19.7.1 Event Filter Parameters

The Event Filter edit page, shown in Figure 19-23, lets you create new and edit existing event filters. To add a new event filter, click add new on the Event Filters list page, or click New or Copy on the Event Filter edit page.

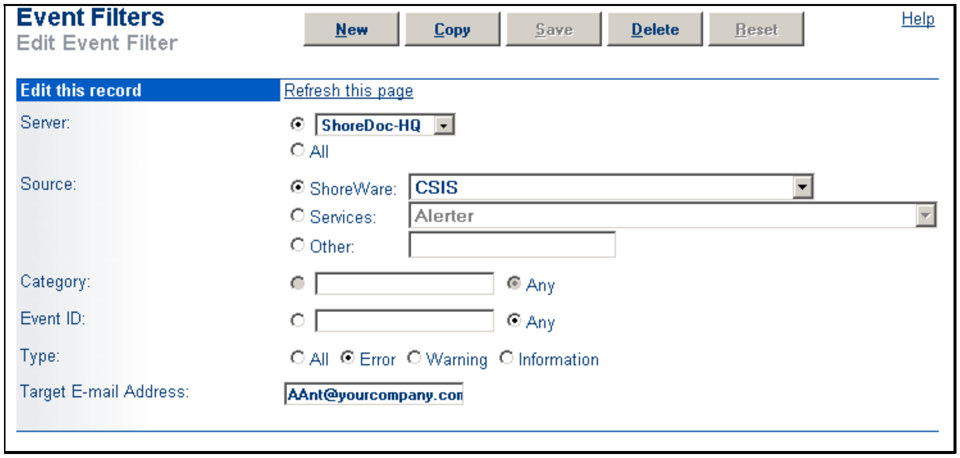


Figure 19-23 Event Filter Edit Page

The parameters on the Event Filter edit are as follows:

- **Server:** This is the server that the event filter runs on. Click **All** or choose a server from the drop-down list.
- **Source:** These options define the event source for the filter. Select from the following:
  - — **ShoreWare—**Notifies about ShoreTel events. The associated drop-down list lets you select the ShoreTel category. Selecting ***Any*** reports events about all ShoreTel categories.
  - — **Services—**Notifies about any non-ShoreTel service.
  - — **Other—**Notifies about any event source.
- **Category:** This lets you select a category when the ***Services*** or ***Other*** option is selected.
- **Event ID:** This lets you select the event identification number that is referenced in the e-mail notification.
- **Type:** These options let you select the severity type:
  - — **Error—**Requires immediate attention.
  - — **Warning—**Alerts to potential errors.
  - — **Information—**Provides status of service, switch, or port.
  - — **All—**Notifies about all three types.
- **Target E-mail Address:** This is the e-mail address of the intended system administration or technical support entity.

19.7.2 Editing and Creating Event Filters

To create or edit an event filter:

Step 1 Open the Event Filters List page by selecting **Maintenance -> Event Filters** in the Navigation frame.

Step 2 To edit an existing filter, click the filter's name in the Event Filters page. To create a new filter, click **Add new**.

Step 3 Click **All** or choose a server from the Server drop-down list.

Step 4 Select a source by clicking the ShoreWare, Services, or Other option.

Sources may have more than one filter associated with them, depending on how you set the criteria. For example, the ShoreTel Event Watch service might have two or more filters, each having different event types assigned to them.

Step 5 Select a source and category based on the following:

- If the ShoreWare option is selected, select an event category from the Category drop-down list or select ***Any*** for all.
- If Services is selected, select a service from the drop-down list, as well as a category.
- If Other is selected, enter a service name, as well as a category.

Step 6 Enter an event number from the Event ID drop-down list or select **Any**.

Step 7 Select an event type by clicking the Error, Warning, Information, or All option.

Step 8 Enter an e-mail address in the Target E-mail Address field.

Step 9 Click **Save** to save the service's filter criteria.

19.8 Database Maintenance

ShoreTel uses MySQL to manage the ShoreTel configuration and CDR databases residing on the Headquarters server. MySQL databases support the UTF-8 character set. All ShoreTel configuration and CDR data is stored in UTF-8 format.

NOTE ShoreTel supports Unicode Character sets by incorporating the v5.1 MySQL ODBC connector, resulting in databases that support Unicode Character sets. Support of Unicode character sets extends to the ShoreTel system and CDR databases. Although Director also supports Unicode Character sets, automatic population of some data entry fields depend on the character set of the originating data.

19.8.1 MySQL Services

19.8.1.1 Monitoring MySQL Service

ShoreTel reports MySQL service status on the Main Server Maintenance window. To access the Main Server Maintenance window, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Maintenance > Quick Look**, the Quick Look page appears.

Step 3 Perform one of the following:

- Click the first site on the list, typically named Headquarters, and click the name of the Soft Switch.
- Click the name of the Main Server, typically named Headquarters Server.

Figure 19-24 indicates the position of the MySQL service monitor on the Main Server Maintenance page. To restart the MySQL service, access the MySQL Command drop down menu on the right side of the page and select Start.

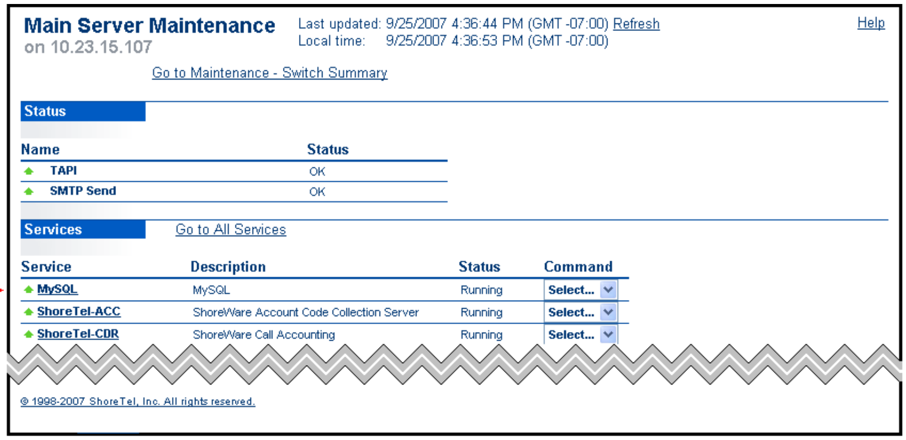


Figure 19-24 MySQL Service Monitor and Command – Main Server Maintenance Page

19.8.1.2 Browsing MySQL Database Tables

MySQL provides a query browser for viewing the database tables. The browser is available at http://dev.mysql.com/downloads/gui-tools/5.0.html

ShoreTel does not support MySQL tools or any problems arise from using these tools.

19.8.1.3 Compatibility with Backup and Antivirus Utilities

Running a virus checker or a backup utility (disk or file) on a MySQL database file causes the MySQL service to crash.

When running these utilities, the MySQL database file (`c:\Shoreline Data\Call Records 2\Data\[ibdata1, ib_logfile*]`) and the folder `C:\windows\temp` must be included in the exclusion list.

19.8.2 MySQL Backup and Restore Utilities

MySQL provides utilities for backing up, restoring, and replicating MySQL databases. The following sections describe the processes for performing these functions.

19.8.2.1 Database Backup

The following command stores a backup copy of the ShoreTel configuration database:

```
(ShoreWare Path)(Command Path)\mysqldump.exe --user=root --
password=shorewaredba --add-drop-database --routines --single-
transaction --port=4308 --databases [database_to_backup]>
[backupfile.sql]
```

where

— *(ShoreWare Path)* is the location where ShoreWare was installed

— *(Command Path)* is the location of mysqldump.exe in the installation directory

— ***backupfile.sql*** specifies the proposed location of the created backup file.

— ***db_name*** is either *shoreware* or *shorewarecdr:*

shoreware is entered to backup the configuration database

shorewarecdr is entered to backup the CDR database

Example: To backup the configuration file where:

- ShoreTel is installed at C:\Program Files\ShoreLine Communications\ShoreWare Server
- mysqldump.exe is located at C:\Program Files\ShoreLine Communications\ShoreWare Server\MySQL\MySQL Server 5.0\bin\
- The backup is to be placed at C:\shoreware_Bkup.sql
- The password is shorewaredba

Execute the following command line:

```
C:\Program Files\Shoreline Communications\Shoreware Server\MySQL\MySQL
Server 5.0\bin\mysqldump.exe --user=root --password=shorewaredba --add-
drop-database --routines --single-transaction --port=4309 --databases
[database_to_backup]> [backupfile.sql]
```

OR change directories, then run the executable:

```
>cd C:\Program Files\Shoreline Communications\Shoreware
Server\MySQL\MySQL Server 5.0\bin\

>mysqldump.exe --user=root --password=shorewaredba --add-drop-database -
-routines --single-transaction --port=4309 --databases
[database_to_backup]> [backupfile.sql]
```

MySQL provides documentation on database backup at:

```
http:/dev.mysql.com/doc/refman/5.0/en/disaster-prevention.html
```

19.8.2.2 Database Restore

The following command restores a backup copy of a ShoreTel configuration database as an active database:

*(ShoreWare Path)**(Command Path)*`\\mysql -u [username] -p [password] [database_to_restore] < [backupfile.sql]`

where

— ***(ShoreWare Path)*** is the location where ShoreWare was installed

— ***(Command Path)*** is the location of the mysqldump.exe within the installation directory

— ***backupfile.sql*** specifies the location of the backup file to be restored.

To restore a configuration file where:

- ShoreTel is installed at C:\Program Files\ShoreLine Communications\ShoreWare Server
- mysql.exe is located at C:\Program Files\ShoreLine Communications\ShoreWare Server\MySQL\MySQL Server 5.0\bin\
- the backup is located at C:\shoreware_Bkup.sql
- the password is shorewaredba

Execute the following command:

```
C:\Program Files\Shoreline Communications\Shoreware
Server\MySQL\MySQL Server 5.0\bin\mysql.exe --user=root --
password=shorewaredba --port=4308 [database_to_restore]<
[backupfile.sql]
```

OR change directories, then run the executable:

```
>cd C:\Program Files\Shoreline Communications\Shoreware
Server\MySQL\MySQL Server 5.0\bin\

>\mysql.exe --user=root --password=shorewaredba --port=4309
[database_to_restore]< [backupfile.sql]
```

19.8.2.3 Database Replication

MySQL provides a built in database replication tool. Refer to the following for information on setting up the database replication:

`http://www.howtoforge.com/mysql_database_replication`

Refer to the following for tools, and add-ons that can assist in database replication:

`http://solutions.mysql.com/solutions/tools?type=25`

The database replication tool is provided by MySQL. ShoreTel does not provide support for MySQL database replication.

19.8.2.4 Database Compacting

When database records are deleted, MySQL does not return the space recovered from the deletion. This results in a database that continues growing as records are added and replaced.

ShoreTel provides a compacting script that recovers unused space that was abandoned when records were deleted. Because the databases are not available during compaction, the script should only be run during scheduled maintenance activities.

To run the MySQL compacting script, open a Windows command prompt and executed the following script:

Program Files\Shoreline Communications\ShoreWare Server\MySQL\MySQL Server 5.0\bin\CompactMySQLData.wsf

The script accepts one optional command line argument – the ShoreTel cdr archive database name. When this argument is not specified, the script uses the default name of shorewarecdrarchive. This argument is required only if the archive database name differs from the default.

CHAPTER 20

System Recovery

This chapter describes methods and tools that protect your system against computer failure. The topics discussed in this chapter include:

20.1 Backup and Restore

ShoreTel recommends that you regularly backup the files on your headquarters server, distributed voicemail (DVM) servers, voicemail switches, and SA-100s. You can use the backed up files to restore the current hardware or upload the files to replacement hardware. A pre-prepared script is available for you to use to preform backup and restore on these ShoreTel devices. The script file is copied to a directory on the ShoreTel headquarters and DVM servers when you install the server software. You can use the script to back up the files listed in Table 20-1.

Table 20-1 Files that Are Backed Up

Headquarters Server	Distributed Voicemail Server	Voicemail Model Switch	Service Appliance 100
/inetpub/mailroot	/shoreline data	/cf	/cf
/intepub/ftproot	/MessageFiles	cfg.dat	cfg.dat
/windows/my.ini	/Prompts	ShoreTel.cfg	ShoreTel.cfg
/shoreline data	/SoftSwitch		
/MessageFiles	/Templates		

Table 20-1 Files that Are Backed Up (Continued)

Headquarters Server	Distributed Voicemail Server	Voicemail Model Switch	Service Appliance 100
/Prompts	/Logs		
/Scripts	/Vms		
/SoftSwitch			
/Templates			
/Call Records 2			
/Database			
/Logs			
/Vms			

The backup and restore script is designed to service a server site. The script backs up and restores only the server it is installed on but can be configured to backup and restore any voicemail switch and SA-100 in your system. You can use a batch file to intiate system-wide backups or restoration. Pre-prepared batch files are also installed in the folder with the backup and restore script that allow you to perform different types of backups. You can use a program such as Microsoft Scheduler to schedule automatic backups.

20.1.1 Estimated Backup and Restore Times

The time it takes to perform a backup or restore depends on many factors including how much information you have to backup or restore. This section provides an estimate of the time it takes to backup or restore different elements in the ShoreTel system for a couple of conditions. Actually times will vary depending on your configuration and environment.

The following is an estimate of time required by ShoreTel backup procedures:

- Clean System (No VM, CDR, or SA-100)
  - Total – 379 secs
  - VM – 2 secs
  - CDR– 35 secs
- Loaded System (VM: 100 messages/13.5 MB; CDR: 500,000 calls)
  - Total – 508 secs
  - Backup VM – 21 secs
  - Backup CDR – 104 secs

The following is an estimate of the time required to restore ShoreWare Server files:

- Clean System
  - Total – 416 secs
  - VM – 3 secs
  - CDR – 32 secs
  - SA-100 –

- Loaded System (VM: 100 messages/13.5 MB – CDR: 500,000 calls)
  — Total – 525 secs
  — VM – 22 secs
  — CDR – 98 secs

20.1.2 Backup Strategy

When backing up a server, server activity must be stopped prior to backing up files to prevent file corruption. The procedures provided by ShoreTel stops server activity before a backup and restarts the server after the backup is complete. ShoreTel recommends backing up files during scheduled down times or periods of light activity.

When backing up an entire system, ShoreTel recommends starting with the distributed voice servers and backing up the headquarters server last. This allows the HQ to remain operational while other servers are unavailable. You can back up multiple distributed voice servers simultaneously. After the distributed voice servers are completed, backup the files on the headquarters server. This operational order minimizes the down time of the headquarters server.

20.1.3 Restoration

The ShoreTel backup and restore script and batch files perform all necessary steps to restore your HQ, DVMs, voicemail switches, and SA-100s. You can use these files to perform a complete restore or a select restore.

Operations and files received on a server after the backup was created are lost when files are restored to the server. When restoring a headquarters server, all files on distributed servers that do not require restoring remain intact; however, voicemail received for mailboxes created since the backup was created may be lost regardless of the server upon which they reside.

Server activity must be stopped while files are restored to a server. The ShoreTel backup and restore script stops the server before the restoring the files and restarts it after the restoration is complete. Files must be restored to a directory identical to the directory in which they were stored.

When restoring an entire system, ShoreTel recommends restoring the headquarters server first so as to establish a functioning system as soon as possible. After the headquarters server is restored, you can restore distributed servers while the headquarters server is active. This operational order minimizes the down time of the headquarters server.

Files can only be restored to the server from which they were backed up. Backup files from the headquarters server can restore only the headquarters server. For systems with more than one distributed server, backup files are not interchangeable between the servers.

20.2 Configuring the Backup and Restore Script

Before you can use the script for backup and restore, you must modify the file with the following information:

- Path of the backup destination folder.
- IP addresses of the voicemail switches for which you want the current server to initiate backup and restore.

- IP addresses of the SA-100 for which you want the current server to initiate backup and restore.
- Drive letter where the script file is installed if changed.
- Path to the script file on the headquarters server or DVM if changed.
- Path on the server where the PLINK and PSCP are found if changed.

To configure the headquarters server and distributed voice servers to perform backup, do the following:

Step 1 On the desktop of the ShoreTel server that you want to use to perform a backup, click **Start > Program Files > Shoreline Communications > ShoreWare Server > Scripts > Sample_Backup_Restore**.

Step 2 Right-click **sw_backup_restore.ini** and select **Edit**. The Window.Install.Drive page appears.

Step 3 On the line "Window.Install.Drive=", type the letter of the drive where the Windows operating system is installed. The default drive is C:.

Step 4 In the Back Options section, do the following to specify where you want to create the backup files:

Step a On the line "Backup.Drive=", type the path for the volume to which you want to back the ShoreTel system files up.

Step b On the line "ShoreWare.Drive=", type the letter of the drive on which the ShoreTel system files you want to backup are stored. The default value is C.

Step c On the line "Backup.Root.Directory =" type the path that you want to use for backing the files up. The default path is:

\Shoreware Backup\Backup

Step d On the line "Backup.Shoreware.Directory =" type the name of the file to which you want to back the files up. The default name is:

\Shoreline Data

Step 5 In the Shoreware File Location section, do the following to specify the location of the ShoreTel files on the current server:

Step a On the line "ShoreWare.Scripts.Root.Directory=", type the path to where the ShoreTel headquarters server backup scripts are located. The default value is:

C:\Program files\Shoreline Communications\ShoreWare Server\Scripts

Step b On the line "ShoreWare.Scripts.DVM.Root.Directory=", type the path to where the ShoreTel DVM server backup scripts are located. The default value is:

C:\Program files\Shoreline Communications\ShoreWare Remote Server\Scripts

Step 6 On the line "VMB.ip.list =", type the IP addresses of the voicemail switches that you want this server to backup. Use a comma to separate the addresses.

Step 7 On the line “UCB.ip.list =”, type the IP addresses of the Service Appliance 100s that you want this server to backup. Use a comma to separate the addresses.

Step 8 On the line “PLINK.CMD =”, type the path where the PLINK command can be found. The default is:

C:\Program files\Shoreline Communications\ShoreWare Server\Scripts\Sample_Backup_Restore\plink

Step 9 On the line “PSCP.CMD =”, type the path where the PSCP command can be found. The default is:

C:\Program files\Shoreline Communications\ShoreWare Server\Scripts\Sample_Backup_Restore\pscp

Step 10 Click **File > Save** to save your changes.

20.3 Preliminary Procedure for Remote Devices

Backup and restore operations are initiated for voice mail switches and SA-100s as remote devices from the HQ or a DVM. To perform backup or restore an SSH connection must be established between the server and the remote device. PLINK and PSCP commands are used to access the remote device using an RSA key for validation. Before you can back up or restore files for a voicemail switch or SA-100, you must cache an RSA key in the registry of the server for each switch or SA-100.

To cache an RSA key for a voicemail switch or SA-100 in a server registry, do the following:

Step 1 Open the Command prompt on the server that you want to use to initiate backup and restore for the switch or SA-100.

Step 2 Type **plink** ***<IP address of the voicemail switch or SA-100>***.

Step 3 Press **Enter**. The program returns a message indicating the storage status of the RSA key in the registry of the server. Figure 20-1 displays the messages if the key is not present.

```
C:\Shoreline Data\Database\Scripts\Sample_Backup_Restore.>plink 10.1.1.242
The server's host key is not cached in the registry. You have no guarantee that the server is
the computer you think it is.
The server's rsa2 key fingerprint is: ssh-rsa 2048
08:b0:43:35:fd:21:1b:c2:6b:27:b4:3f:a9:f7:be:2d
If you trust this host, enter "y" to add the key to PuTTY's cache and carry on connecting.
If you want to carry on connecting just once, without adding the key to the cache, enter "n".
If you do not trust this host, press Return to abandon the connection.
Store key in cache? (y/n) y
login as: ^C
```

Figure 20-1 Caching the Registry Key By Using Plink.

Step 4 If the key is not cached to the registry, press **y** when prompted.

Step 5 Repeat Steps 2 through 4 for each remote device for which you want this server to initiate backup and restore operations.

Step 6 Save and close the file.

20.4 Backing Up the Headquarters Server

This section describes how to backup your headquarters servers. It describes how do a complete backup of the server and a partial backup of the server.

NOTE Server activity must be stopped before backing up files to prevent corruption caused from modifying files during its backup. The processes provided by ShoreTel stop the server before the backup and restarts the server after the backup is complete. ShoreTel recommends backing up files during scheduled down times or periods of light activity.

20.4.1 Performing a Complete Backup

To perform a complete backup of your headquarters server, do the following:

Step 1 Access the command prompt on the headquarters server.

Step 2 Navigate to the directory where the ShoreTel backup and restore scripts are found. The default path is:

c:\program files (x86)\shoreline communications\shoreware server\scripts\sample_backup_restore

Step 3 At the prompt, type:

cscript.exe shoreware_backup.wsf hq all

Press Enter.

20.4.2 Performing a Selective Backup

You can perform a backup of selected files. To use the headquarters server to backup selected files, do the following:

Step 1 Access the command prompt on the headquarters server.

Step 2 Navigate to the directory where the ShoreTel backup and restore scripts are found. The default path is:

c:\program files\shoreline communications\shoreware server\scripts\sample_backup_restore

Step 3 At the prompt, type:

cscript.exe shoreware_backup.wsf ***x y***

where ***x*** is component type that you want to backup and ***y*** is the file type as shown in Table 20-2.

Press **Enter**.

Table 20-2 Backup Arguments

Component		File Type	
Argument	**Definition**	**Argument**	**Definition**
hq	Headquarters server	db	database (not valid if the server_type = dvm or vmb)
dvm	Distributed Voice Mail server	vm	voicemail (not valid if server type = vmb)
vmb	Voicemail Model Switch	log	log files (not valid if server type = vmb)
ucb	Service Appliance 100	ucb	conference bridge

20.5 Backing Up Distributed Voice Mail Servers

This section describes how to backup your distributed voicemail servers. It describes how do a complete backup of the server and a partial backup of the server.

NOTE Server activity must be stopped before backing up files to prevent corruption caused from modifying files during its backup. The processes provided by ShoreTel stop the server before the backup and restarts the server after the backup is complete. ShoreTel recommends backing up files during scheduled down times or periods of light activity.

20.5.1 Performing a Complete Backup of a DVM

To perform a complete backup of a DVM, do the following:

Step 1 Access the command prompt on the DVM that you want to backup.

Step 2 Navigate to the directory where the ShoreTel backup and restore scripts are found. The default path is:

c:\program files\shoreline communications\shoreware server\scripts\sample_backup_restore

Step 3 At the prompt, type:

cscript.exe shoreware_backup.wsf hq all

Press **Enter**.

20.5.2 Performing a Selective Backup of a DVM

You can perform a backup of selected files. To use the headquarters server to backup selected files, do the following:

Step 1 Access the command prompt on the DVM that you want to backup.

Step 2 Navigate to the directory where the ShoreTel backup and restore scripts are found. The default path is:

c:\program files\shoreline communications\shoreware server\scripts\sample_backup_restore

Step 3 At the prompt, type:

cscript.exe shoreware_backup.wsf ***x y***

where ***x*** is component type that you want to backup and ***y*** is the file type as shown in Table 20-2 on page 575.

Press **Enter**.

20.6 Backing Up Voice Mail Switches

Backup of voice mail switches must be initiated on the headquarters server or a DVM. The backup script on the server must be modified to include the IP addresses of the voice mail switches that you want to backup (see the "Configuring the Backup and Restore Script" on page 571 for information about modifying the script file). This section describes how to backup voice mail switches.

Requirements

- Headquarters server or DVM to implement backup.
- sw_backup_restore.ini file on implementation server modified to include IP address of voice mail switch.
- Server able to establish an SSH connection with the voicemail switch.

To backup voice mail switches, do the following:

Step 1 Access the command prompt on the ShoreTel server that is configured to backup to the voice mail switches you want to backup.

Step 2 Navigate to the directory where the ShoreTel backup and restore scripts are found. The default path is:

c:\program files\shoreline communications\shoreware server\scripts\sample_backup_restore

Step 3 At the prompt, type:

cscript.exe shoreware_backup.wsf vmb all

Press **Enter**.

20.7 Backing Up Service Appliance 100s

Backup of SA-100s must be initiated on the headquarters server or a DVM. The backup script on the server must be modified to include the IP addresses of the SA-100s that you want to backup (see the "Configuring the Backup and Restore Script" on page 571 for information about modifying the script file). This section describes how to backup SA-100s.

Requirements

- Headquarters server or DVM to implement backup.
- sw_backup_restore.ini file on implementation server modified to include IP address of voice mail switch.
- Server able to establish an SSH connection with the SA-100.

To backup SA-100s, do the following:

Step 1 Access the command prompt on the ShoreTel server that is configured to backup to the SA-100s you want to backup.

Step 2 Navigate to the directory where the ShoreTel backup and restore scripts are found. The default path is:

c:\program files\shoreline communications\shoreware server\scripts\sample_backup_restore

Step 3 At the prompt, type:

cscript.exe shoreware_backup.wsf ucb all

Press **Enter**.

20.8 Restoring the Headquarters Server

This section describes how to use the ShoreTel backup and restore script to perform complete and selective restores to the HQ.

20.8.1 Performing a Complete Restore

To perform a complete restore of your headquarters server, do the following:

Step 1 Access the command prompt on the headquarters server.

Step 2 Navigate to the directory where the ShoreTel backup and restore scripts are found. The default path is:

c:\program files\shoreline communications\shoreware server\scripts\sample_backup_restore

Step 3 At the prompt, type:

cscript.exe shoreware_restore.wsf hq all

Press **Enter**.

20.8.2 Performing a Selective Restore

You can perform a restore of selected files. To use the headquarters server to restore selected files, do the following:

Step 1 Access the command prompt on the headquarters server.

Step 2 Navigate to the directory where the ShoreTel backup and restore scripts are found. The default path is:

c:\program files\shoreline communications\shoreware server\scripts\sample_backup_restore

Step 3 At the prompt, type:

cscript.exe shoreware_restore.wsf *x y*

where ***x*** is component type that you want to backup and ***y*** is the file type as shown in Table 20-2.

Press **Enter**.

Table 20-3 Restore Arguments

Component		File Type	
Argument	Definition	Argument	Definition
hq	Headquarters server	all	
dvm	Distributed Voice Mail server	db	database (not valid if the server_type = dvm or vmb)
vmb	Voicemail Model Switch	vm	voicemail (not valid if server type = vmb)
ucb	Service Appliance 100	ucb	conference bridge
		log	log files (not valid if server type = vmb)

20.9 Restoring Distributed Voice Mail Servers

This section describes how to use the ShoreTel backup and restore script to perform complete and selective restores to DVMs.

20.9.1 Performing a Complete Restore of a DVM

To perform a complete restore of a DVM, do the following:

Step 1 Access the command prompt on the DVM that you want to restore.

Step 2 Navigate to the directory where the ShoreTel backup and restore scripts are found. The default path is:

c:\program files\shoreline communications\shoreware server\scripts\sample_backup_restore

Step 3 At the prompt, type:

cscript.exe shoreware_restore.wsf hq all

Press **Enter**.

20.9.2 Performing a Selective Restore of a DVM

You can perform a restore of selected files. To use the headquarters server to restore selected files, do the following:

Step 1 Access the command prompt on the DVM that you want to restore.

Step 2 Navigate to the directory where the ShoreTel backup and restore scripts are found. The default path is:

c:\program files\shoreline communications\shoreware server\scripts\sample_backup_restore

Step 3 At the prompt, type:

cscript.exe shoreware_restore.wsf ***x y***

where ***x*** is component type that you want to backup and ***y*** is the file type as shown in Table 20-2 on page 575.

Press **Enter**.

20.10 Using Batch Files

ShoreTel has created batch files that you can use to backup your ShoreTel components. The batch files are located in the same folder as the backup-and-restor script which they use to perform backups. These batch files allow you to backup or restore different ShoreTel components on the site including the following:

- Headquarters.
- Distributed voice mail.
- Voicemail Model Switches.
- SA-100s (UCBs).

To batch file commands for backup are as follows:

- hq_backup_all.bat
- dvm_backup_all.bat
- vmb_backup_all.bat
- ucb_backup_all.bat

The batch commands for restore are as follows:

- — hq_restore_all.bat
- — dvm_restore_all.bat
- — vmb_restore_all.bat
- — ucb_restore_all.bat

To use a batch file to backup or restore files, do the following:

Step 1 Access the command prompt on the ShoreTel server.

Step 2 Navigate to the directory where the ShoreTel backup and restore scripts are found. The default path is:

c:\program files (x86)\shoreline communications\shoreware server\scripts\sample_backup_restore

Step 3 At the prompt, enter the batch file that you want to use for backup or restore. Press Enter.

20.10.1 Log Files

Log files display the commands that are performed during Backup and Restore operations at SwBackupRestore.log. By default, Windows maintains three log files.

20.11 Failover Support

To provide high availability, ShoreTel supports failover at two critical points in the system structure. ShoreTel supports failover for the headquarters server and for voice switches. For the headquarters server, you can install a backup server that mirrors and monitors the primary server. If the primary server fails, operations are immediately transferred to the backup server with minimal interruption of telephony services. After the primary server is repaired, you must manually failback the secondary server which returns operations to the primary server and returns the backup server to the monitoring and backup role.

For switches you build in failover by setting up your system with extra port capacity or installing spare switches that can be activated within minutes to temporarily take on the phones that lose connection with the failed switch. Spare switches can be physically installed on a remote network from the failed switch and its phones, and may not provide the same level of service.

This section discusses failover at the server level. For more information about switch failover, see the "Failover for IP Phones: Spare Switch" on page 103.

To provide failover protection for ShoreTel servers, ShoreTel recommends that you use Double-Take. For complete information on implementing Double-Take failover protection, refer to the Double-Take application note available on the ShoreTel Website.

20.11.1 Configuring a Secondary IP Address

After you have created a backup server for your system, you must designate the backup server for failover function. To designate a backup server for the failover function, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Application Servers**. The Application Servers page appears.

Step 3 Select the Headquarters Server. The Edit Server page for the Headquarters server appears as shown in Figure 20-2.

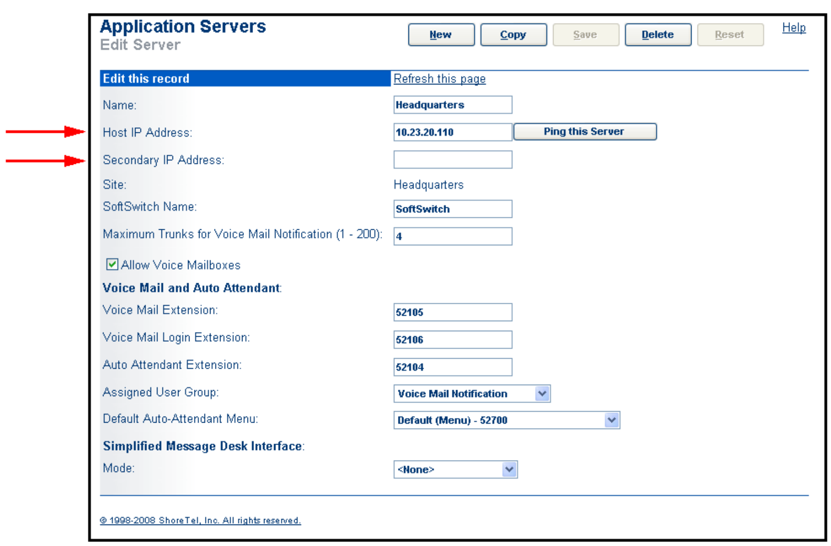


Figure 20-2Headquarters Edit Server Page

Step 4 In the Secondary IP Address field, enter the IP address of the ShoreTel server that you want to use for failover.

To convert a system to single Headquarters Server mode, remove the address from the secondary IP address field, then reboot the Headquarters server.

When the primary and secondary servers are in different subnets, their IP addresses must be static.

The DNS server accessed by a ShoreTel system must associate the same server name to the primary and secondary servers if:

- The primary and secondary servers are configured with static IP addresses.

  and

- The HQ server supports Communicator clients or clients on Citrix or Windows Terminal servers.

The primary and secondary Headquarters server cannot be running and connected to the network at the same time.

20.11.2 Conditions during the Failover and Failback Operations

ShoreTel performs a failover operation when the primary server fails to transfer Headquarters server functions to the secondary server. After the failover operation is complete, the secondary server performs all Headquarters server functions.

The administrator can initiate a failback operation to restore Headquarters server function to the primary server. After the failback is complete, the system structure that existed before the failback operation is restored.

Failover and failback operations typically require between five to 20 minutes, depending on the system configuration. During these periods, no Headquarter server services are available. Failover and failback operation effects are resolved after the operations are complete and the secondary (for failover) or primary (failback) server is functioning as the Headquarters server.

Failover and failback operation effects on Communicator include:

- Users configured on the Headquarters server lose most connectivity capabilities, including telephony and video. IM connectivity is not affected.
- Users configured on distributed servers maintain all connectivity capabilities.
- Configuration changes are unavailable.
- Users logging into Communicator while the secondary server controls the system must specify the IP Address of the secondary server.

  Refer to the Communicator User's Manual for instructions on specifying the server IP address.

Failover and failback operation effects on voicemail include:

- Sites that receive voicemail through the Headquarters server lose voicemail access.
- Mailboxes lose access to any voicemail that is routed through the Headquarters server.
- Sites that receive voicemail from distributed servers retain voicemail access.

Failover and failback operation effects on other system components include:

- Account Code and Workgroup services are not available.
- Director, Web Client, and Communicator configuration changes are not available.
- If DRS is enabled, intersite calls are unavailable to users residing on sites whose only access to a DRS server is through the Headquarters' DRS.

20.11.3 System Failover Conditions and Requirements

After the failover operation is complete, the secondary server performs all distributed server and application connectivity activities managed by the Headquarters server. The following sections describe required administrator tasks after the failover operation.

20.11.3.1 License Compliance

After a failover operation transfers Headquarters server control to the secondary server, license status on the secondary server is non compliant. To restore the system to compliance, reinstall all licenses that were originally purchased for the secondary server.

20.11.3.2 Communicator Failover Requirements

To assure proper Communicator client operation and performance after a Failover or a Failback, the browser cache must be cleared. The system administrator should send messages to all MCM clients after a failover operation advising them to clear the browser cache on their devices.

20.11.4 System Failback Conditions and Requirements

After the failback operation is complete, the primary server resumes all distributed server and application connectivity activities managed by the Headquarters server. The following sections describe required administrator tasks after the failback operation.

20.11.4.1 License Compliance

After a failback operation transfers Headquarters server control to the secondary server, license status on the secondary server is non compliant. To restore the system to compliance, reinstall all licenses that were originally purchased for the secondary server.

20.11.4.2 MCM Failback Requirements

To assure proper MCM client operation and performance after a failback, the browser cache must be cleared. The system administrator should send messages to all MCM clients after failback procedures advising them to clear the browser cache on their devices

20.12 Failover and Restoration of IP Phones

When you enable the system parameter to failover IP phones, IP phones will automatically failover to another switch on the same site or to a spare switch when they lose connection with their primary switch. The spare switch is designed as a temporary measure to ensure that IP phone users have basic phone connectivity should their primary switch fails. To ensure that users have their full connectivity, you must repair or replace the failed primary switch as soon as possible. This section describes how to restore normal operation after a failover occurs. This discusses the following topics:

- "Re-assigning the Primary Switch Profile to a Replacement Switch" on page 583.

- "Moving IP Phones to Primary Switch" on page 583.
- "Failing Back the Spare Switch" on page 584.
- "Verifying Spare Switch Return Status" on page 585.

20.12.1 Re-assigning the Primary Switch Profile to a Replacement Switch

If you must physically replace a primary switch that fails, you can re-assign the original switch profile to the new physical switch rather than create a new profile. This section describes how to re-assign the switch profile.

Requirements:

- Obtain a replacement switch that has the same capabilities as the failed switch.
- Physically install the replacement switch on the same network as the old switch.
- Assign the new switch an IP address. Refer to the x for more information about.
- Unplug the port connections (telephones, trunks) from the existing voice switch and plug them into the new voice switch.

To re-assign the switch profile:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Platform Hardware > Voice Switches/Service Appliances > Primary**. The Primary Voice Switches/Service Appliances page appears.

Step 3 Select the voice switch that is being replaced. The Edit Switch page for the switch appears.

Step 4 In the IP Address field, enter the IP address (or Ethernet address) of the new switch that you want to use to replace the downed switch.

NOTE You can also use the Find Switch button.

Step 5 Select the new voice switch and click **Save**. It can take up to two minutes for the switch to come on line.

NOTE You can use Quick Look to confirm that the new voice switch is on line.

20.12.2 Moving IP Phones to Primary Switch

To move IP phones from the spare switch to the primary switch, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > IP Phones > Individual IP Phones**. The IP Phones page appears as shown in Figure 20-3.

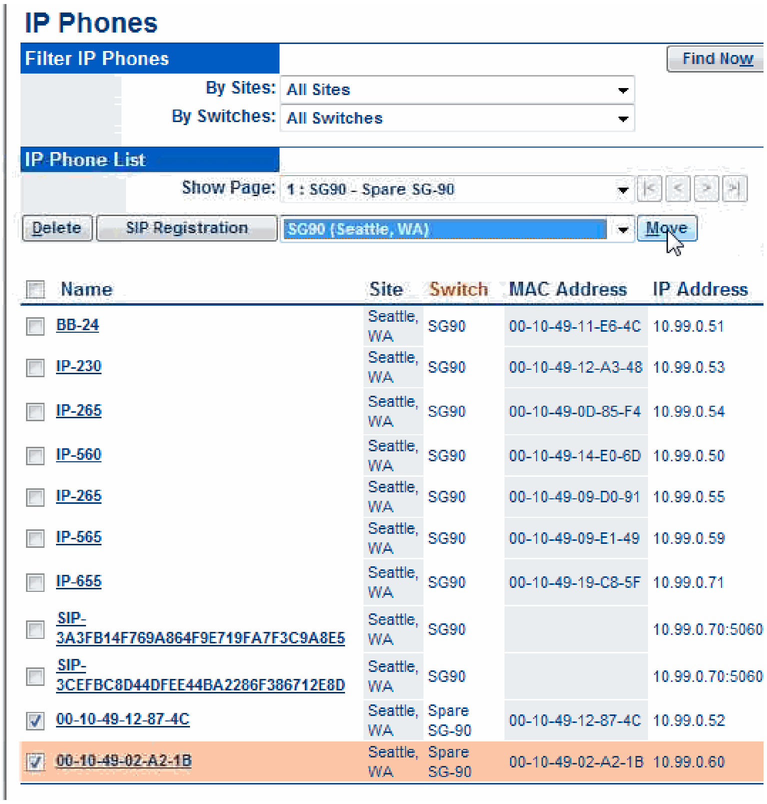


Figure 20-3IP Phone Page

Step 3 In the By Sites field, select the site where the failover has occurred and you want to perform restoration.

Step 4 In the Use the Switches field, select All Switches.

Step 5 In the Show Pages field, select the page that contains the IP phones that you want to move. The first name and the last name of the phones listed on the page are shown in the field along with the page number.

Step 6 Use the Site column to identify phones that have failed over to the spare switch that you want to move to the primary switch and check the check box to the left of the phone names (the MAC or IP address are often used as name).

NOTE You can select multiple phones to move at one time. The phones do not have to be registered to the same switch.

Step 7 In the field to the left of the Move button, select the switch to which you want to move the IP phones.

Step 8 Click **Move**. The phones are moved to the target primary switch.

NOTE Calls that are currently in progress are dropped during the move.

20.12.3 Failing Back the Spare Switch

After you move the IP phones to the primary switch on the site, you must manually failback the spare switch. To failback the spare switch, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Maintenance > Quick Look**. The Quick Look page appears.

Step 3 Select the site where the failover occurred and to which the spare switch is currently assigned. The Voice Switches and Service Appliances Summary page appears similar to Maintenance Summary page shown in Figure 20-4.

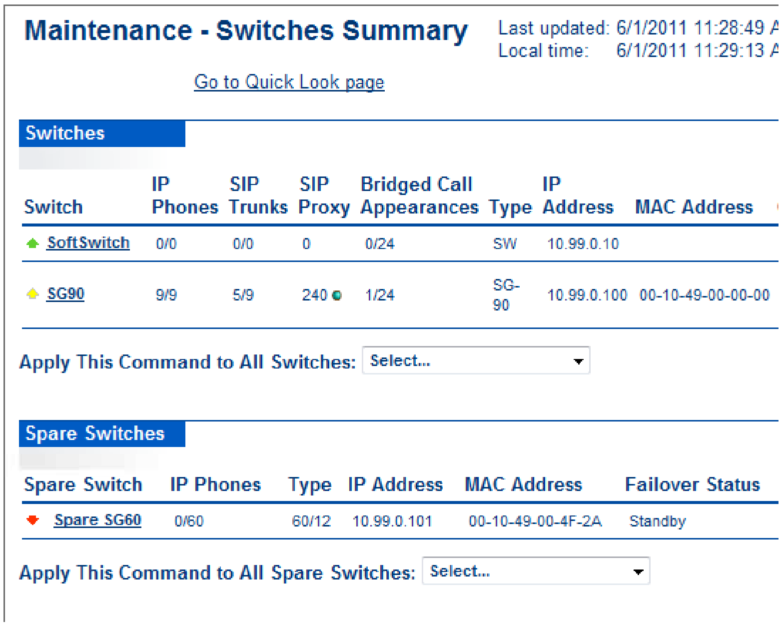


Figure 20-4Maintenance Summary Page

Step 4 In the Spare Switches section, identify the spare switch to fail back and in the Command column field, select **Failback**. The failback process starts.

> **NOTE** Make sure that zero (0) IP phones are connected to the switch. (The listing in the IP Phones column should be 0/N where 0 is the number of phone currently registered with the switch and N is the switch capacity.)

The process takes a few minutes to complete and includes rebooting the spare switch. When the process is complete and successful, the spare switch returns to the spare state.

20.12.4Verifying Spare Switch Return Status

To verify that the switch has returned to the spare state, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Platform > Voice Switches/Service Appliances > Spare**. The Spare Voice Switches page appears.

Step 3 Verify the following columns:

- The Current Site column is empty.
- The IP Phones in Use column lists zero (0).

CHAPTER 21

Reporting

ShoreTel Director allows you to generate reports from information gather throughout the ShoreTel system. You can generate activity reports about several topics as Call Detail Record (CDR) and about Web conferences. You can also set management parameters for handling reports such as setting up alternative report output, enabling archiving, and setting language parameters. This chapter provides information about the reporting capabilities of the ShoreTel system. The topics discussed include:

- "Accessing the Call Details Reports Page" on page 587.
- "Web Conference Reports" on page 589.
- "Options" on page 590.
- "Supporting Asian Fonts" on page 591.
- "Configuring Send CDR Out SMDR Interface" on page 592.

21.1 Accessing the Call Details Reports Page

To access the CDR page, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Reports > Call Details**. The Call Details Report page appears as shown in Figure 21-1. This page is the primary method of accessing and viewing the CDR data in the MySQL database.

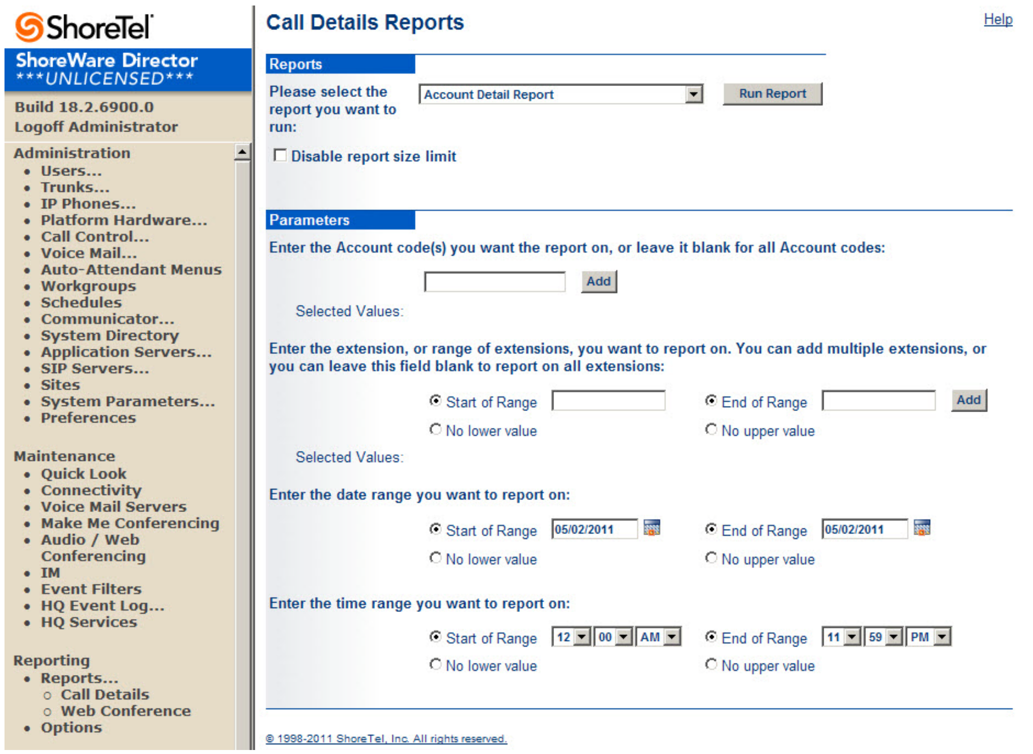


Figure 21-1 Call Details Report Page

Step 3 In the **Please select...** field, select the report that you want to run.

Reports can be run from ShoreTel Director. After the report has been generated, it can be printed, exported, and navigated interactively, similar to compiled reports.

NOTE No more than two users should run reports at the same time. Having more than two people generating reports simultaneously can adversely impact system performance.

21.1.1 Call Details Reports Page

21.1.1.1 Reports section

- **Please select the report you want to run:** This drop-down lists all of the different types of reports that can be run. Options are:
  - — Account Detail Report
  - — Account Summary Report
  - — Media Stream Detail Report
  - — Media Stream Summary Report
  - — Trunk Activity Detail Report
  - — Trunk Activity Summary Report
  - — Workgroup Queue Summary Report
  - — Workgroup Service Level Summary Report
  - — User Activity Detail Report
  - — User Activity Summary Report
  - — Workgroup Agent Detail Report
  - — Workgroup Agent Summary Report

Refer to Appendix B for more information.

21.1.1.2 Parameters Section

- **Enter the account code(s) you want to report on:** Account codes are typically used to assist ShoreTel users in the billing of their clients (for example, a law firm tracking the length of a call). Enter the account code(s) that corresponds to one or more clients in the field provided and click the **Add** button to enter the number.
- **Enter the extension, or range of extensions, you want to report on:** Enter a single extension or a range of extensions in the fields provided to generate a report on only those extensions. You can also leave the fields blank to run the report on all extensions. You can enter an upper or lower extension and then select the **No lower value** or the **No upper value** radio button and to run a report on all extensions up to a specified extension.
- **Enter the date range you want to report on:** Click on the Calendar icon to select a **Start of Range** date and/or a **End of Range** date in the fields provided to generate a report on only those extensions.

  Select the **No lower value** radio button and the **No upper value** radio button to run a report on run a report on all valid dates for which the system has CDR data.
- **Enter the time range you want to report on:** Click on the **Start of Range** radio button and use the drop-down menus to select the start time for the report. Click the **End of Range** radio button and use the drop-down menus to select the outside time limit for the report.

  Select the **No lower value** radio button and the **No upper value** radio button to run a report on run a report on all hours within the specified dates.

21.2 Web Conference Reports

Director can also be used to generate Web Conference Reports from a local host or a remote server.

To generate Web conference reports do, the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Reporting > Reports > Web Conference**. The Web Conference Report page appears as shown in Figure 21-3.

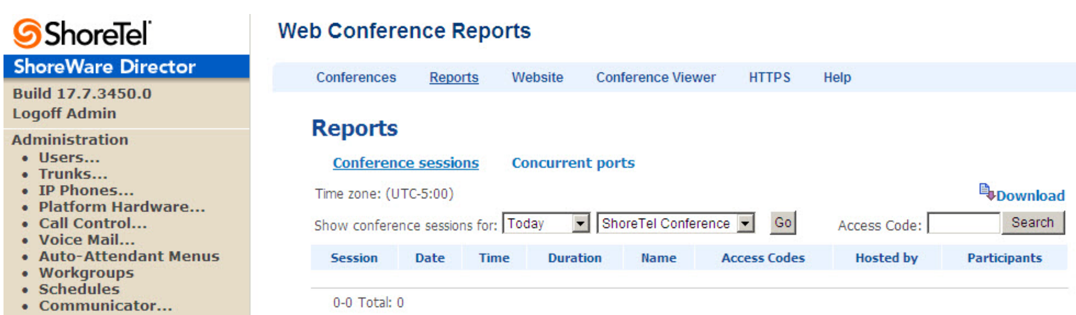


Figure 21-2 Web Conference Reports Page

Reports can be generated based on the date of the conference, the type of device (conference bridge or Service Appliance), or the access code assigned to scheduled conferences.

NOTE This process may cause some implementations of Microsoft Internet Explorer 8 to fail. If downloading Web report files causes your implementation of Microsoft IE8 to fail, enable the IE8 security parameter "**Websites in less privileged web content zone can navigate into this zone.**" To navigate to this parameter in IE 8, click **Tools > Internet Options >Security Tab >Local Intranet > Custom Level**. Scroll down the page to the parameter and click **Enable**.

21.3 Options

The Reporting Options page as shown in Figure 21-3 is accessed by clicking on the Options link under Reporting.

The ShoreTel system supports the ability to send CDR data out a serial port on the main ShoreTel server. The Reporting Options page allows a system administrator to designate which COM port to enable. CDR data is subsequently sent out this port (in addition to being sent to the regular text file and/or a database). Sending the CDR data out the serial port does not change the formatting.

Figure 21-3 Reporting Options Page

The Reporting Options Page is described as follows:

- **Refresh this page:** Click this link to update the information displayed on the page. The **Reporting Options** page automatically updates every 60 seconds.
- **COM Port for CDR Output:** Select the desired COM port on the headquarters server when CDR data will be sent. The default value is none, with a valid range of 1 – 10.
- **Retention Period for CDR Archive:** This value ranges from 1 day to 2000 days. Default value is 125 days.
- **Retention Period for CDR Data:** This value ranges from 1 day to 2000 days. Default value is 125 days.
- **Enable Archiving:** Select this check box to enable archiving.
- **Archive Database Name:** Enter the name of the archive database. Saving the name of the database will not create the archive database; a separate utility must be used.

See the "Creating the Archive Database" on page 593 for details on creating an archive database.

- **Archive Database IP Address:** Enter the IP address of the server where the archive database is stored.
- Select Language Variant: The field specifies the Asian Font that is supported on the computer running Director. Refer to "Supporting Asian Fonts" on page 591 for more information.

21.4 Supporting Asian Fonts

Ascender Corporation provides four font files (Arial Unicode for ShoreTel, Arial Unicode for ShoreTel Bold, Arial Unicode for ShoreTel Italics, and Arial Unicode for ShoreTel Bold Italics) for three languages: Japanese, Simplified Chinese, and Traditional Chinese. Only one set of files – supporting one language – can be installed on a computer at any time.

To install a font on a machine where web reports are required, perform the following:

Step 1 Open the Edit Reporting Options page by selecting **Reporting -> Options** from the Directory Main Menu. Figure 21-4 displays the Edit Reporting Options page.

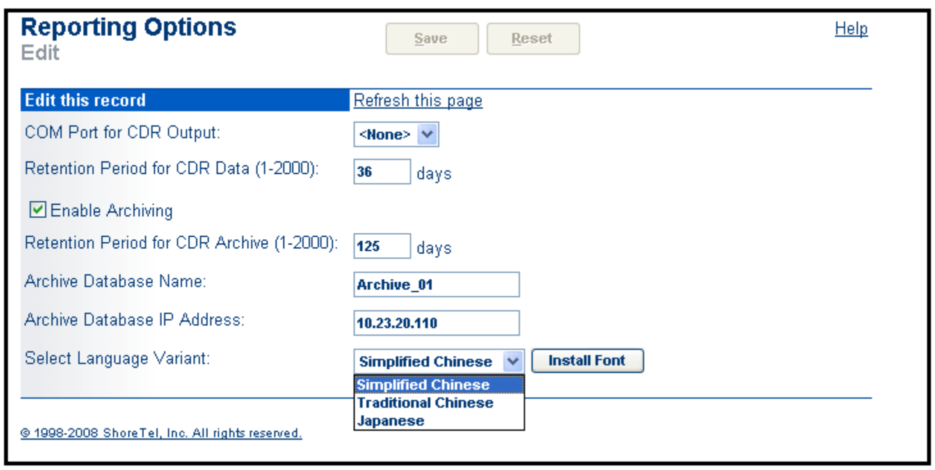


Figure 21-4 Reporting Options Page

Step 2 Select the desired language from the Select Language Variant drop-down menu.

Step 3 Press the Install Fonts button. ShoreTel installs the requested font set on the computer.

Chinese Traditional font is installed on the Headquarters server by default. Access reports from a client machine or using a different Asian font requires a font installation through the Reporting Options page.

21.5 Configuring Send CDR Out SMDR Interface

The ShoreTel system captures CDRs in a database in a text-file format. However, for legacy call accounting systems that cannot read CDR from a database, the CDR data can be delivered as a Station Messaging Detail Record (SMDR) using a serial (COM) port on the main ShoreTel server. When using SMDR, the following applies:

- Formatting of the CDR data remains the same, regardless of whether it is sent out the COM port or written to the database.
- The application should auto-detect the serial port configuration by extracting information about the status of the serial port configuration (e.g. baud rate) from the Windows registry.
- The feature will be disabled by default and must be enabled by selecting a COM port.
- If the serial port should become unavailable through an event such as becoming locked by extremely high volumes of traffic, the CDR data will be queued in a buffer for 300 seconds to help prevent the loss of data. If the serial port returns to service within the 300-second time period, the streaming will resume.

To designate a COM port in order to send CDR Out SMDR Interface via ShoreTel Director:

Step 1 Launch ShoreTel Director.

Step 2 Click **Reporting > Options**. The Reporting Options page appears.

Step 3 Click **COM Port.** The CDR Output drop-down menu appears as shown in Figure 21-5.

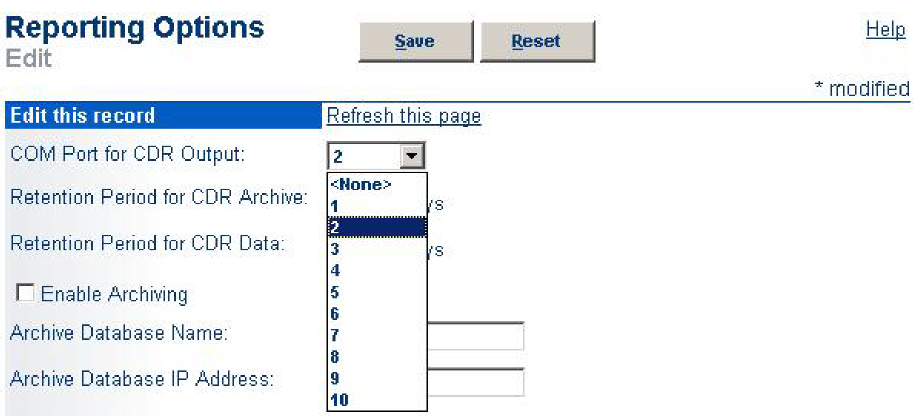


Figure 21-5 Selecting the desired COM port

Step 4 Select the COM port that you want to use for SMDR on the headquarters server.

Step 5 Click **Save** to store your changes.

After the COM port is configured, the system directs CDR data to the SMDR port as calls move through the system.

21.5.1 Creating the Archive Database

The following procedure describes the process of creating a separate (optional) archive database using the MakeCDRArchive.exe command line utility.

Step 1 On the ShoreTel headquarters server, navigate to *C:\Program Files\Shoreline Communications \ ShoreWare Server* and make sure the following files are installed in the same directory:

- MakeCDR.dll
- MakeCDR.sql
- MakeCDR_sp.sql

Step 2 Open the command prompt window in the directory shown above and run the following command:

MakeCDRArchive -s localhost -d databasename

- Databasename is the name of the archive database to be created. The database name must be the same as the name created on the **Options** page.
- If no name is defined, the default name of *shorewarecdrarchive* will be created within the following directory: *C:\Shoreline Data \ Call Records2 \ Data*

Records from main database will be archived to the archive database when the services start up and every night at approximately 12 a.m.

NOTE Instructions for installing MySQL on a secondary server is provide in the "MySQL Database" on page 616

APPENDIX A

Emergency Dialing Operations

This chapter explains the chain of events in the call flow when a emergency call is placed. This chapter also provides instructions for configuring your ShoreTel system to ensure that emergency services are dispatched to the correct location. And finally, the chapter tells you how to select which of the various pieces of caller ID information will be used to identify callers when an emergency call is placed. The topics discussed in this chapter include:

A.1 How Emergency Calls Work

This section provides a simple scenario of how emergency calls are handled with the ShoreTel system. Figure A-1 displays a simple emergency call-flow scenario.

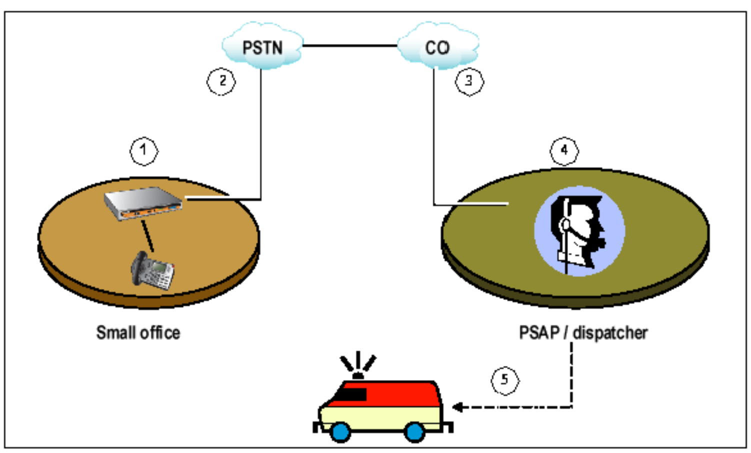


Figure A-1 Simplified Emergency Call Flow Scenario

A.1.1 Emergency Call Scenario

The following is a description of the call flow depicted in the figure.

1. A emergency call is placed from a ShoreTel desk phone.
2. The ShoreTel system identifies the call as an emergency and automatically routes it to an outbound trunk. Caller ID information is provided in either of the following ways:
   - When the call is sent over a PRI trunk, the ShoreTel system provides caller ID information.
   - When the call is sent over a non-PRI trunk, the service provider provides caller ID information.
3. The call is passed over the Public Switched Telephone Network (PSTN) to the exchange of the service provider.
4. The service provider passes the call to a Public Safety Answering Point (PSAP). This is the location of the emergency services dispatcher.
5. The dispatcher at the PSAP gets a “screen pop” which displays information contained in a emergency database. The database contains a mapping between the caller ID number and the geographic location of the caller.
6. The dispatcher sends emergency response personnel to the calling party’s location.

For emergency calls placed from residential or a single-site businesses, determining the location of the calling party is fairly simple and straightforward. However, when dealing with large offices and campus environments, your emergency configuration can get complex. If you are maintaining a configuration that has many remote sites, it is imperative that you do the following:

- Keep your emergency information current with your PSAP.
- Work with your service provider to find out what kinds of caller ID information they will accept.
- Work with the local PSAP to ensure that any changes in your emergency configuration (i.e. names, phone numbers, locations of the members) are mirrored in the PSAP’s database.

A.1.2 Roles and Responsibilities

Each participant in an emergency call has a different role to fill and a different set of responsibilities to handle.

The role of the PBX is to:

- Identify the call as an emergency call.
- Route the call to an outbound trunk. (Preferably a dedicated trunk.)
- Pass the correct caller ID information to the exchange of the service provider when a PRI trunk is used to send the call.

The role of the exchange of the service provider is to:

- Work with the customer to ensure the correct caller ID number is passed to the PSAP.
- Pass the caller information to the PSAP.

  NOTE The billing number of the trunk is used if no other caller information is available.

The role of the PSAP is to:

- Receive emergency calls.
- Host a database that maps the caller ID numbers to the physical location of the users.
- Display information about the calling party to a dispatcher.
- Send the proper emergency response personnel to the caller's location.

The role of the customer (i.e. you) is to:

- Decide which type of caller ID information best fits your needs for emergency calls.
- Work with the service provider to verify that they will accept your preferred type of caller ID information.
- Communicate any changes to your emergency configuration to ensure the PSAP is current.

A.2 Using a PS/ALI Service Provider

In addition to working with your local PSAP to provide accurate logistical information, we recommend that you subscribe to a Private Switch/Automatic Location Information (PS/ALI) service provider as well.

A PS/ALI service provider maintains a database that stores specific address information for each extension or DID on your system. Subscribing to PS/ALI services ensures that accurate automatic number identification (ANI) information is passed to the PSAP in the event of an emergency call, and prevents the emergency responder from showing up at the wrong location.

A subscription to a PS/ALI service provider is particularly recommended in situations where a ShoreTel system is deployed in an environment where a single PRI is used to serve multiple locations (for example, if a single PRI is used for several schools in the same district). In such environments, it is possible for a user to make an emergency call from one of the elementary schools and have the emergency crews dispatched to the wrong location. This can happen if the local trunks are busy and the call gets routed across an analog trunk and across the WAN to the first available PRI, which might be at one of the other schools in the district. With no PS/ALI database to provide accurate information about the origination of the call, the emergency services providers see the call originating at the wrong location.While the correct phone number is sent to emergency services, the association is with the PRI instead of the school where the call originated.

This critical error can be prevented if a PS/ALI database is in place. Such a database (maintained by a PS/ALI service provider) can identify the location associated with a specific DID.

NOTE ShoreTel does not provide PS/ALI service. Contact the local telco carrier for information about PS/ALI service providers in the relevant areas.

A.3 Feature Operation

A.3.1 Digit Collection for Emergency Calls

A ShoreTel user who dials an emergency number (or <access_code> + emergency number) will be routed to an emergency-capable trunk.

- If the user dials an access code followed by an emergency number, digit collection terminates immediately and the call is routed to an emergency-capable trunk.
- If the user forgets to dial an access code before dialing the emergency number, the system waits five seconds before routing the call to an emergency-capable trunk. This pause has been introduced to eliminate accidental calls to the emergency number.

  NOTE Systems that use 911 for the emergency number often also use 9 as an access code for outbound calls. This makes it easy for users to mistakenly dial 911 on a long-distance call by adding an extra 1 before the area code (for example, if he or she dialed the following number 9-1-1-408-555-1212). If additional digits are entered after 9-1-1 during the five-second timeout period, the system will consider it a dialing error and the calling party will hear a reorder tone.

A.3.2 Ensuring Proper Routing of Emergency Calls

Without a dedicated emergency-enabled trunk, emergency calls may not be properly routed under the following circumstances:

- If all available emergency-enabled trunks are busy, the ShoreTel system will not route the emergency call.
- If a site has no emergency-enabled trunk and 'Parent as Proxy' is enabled for that site, the ShoreTel system will not route the call to the emergency-enabled trunks of the parent site if the admission control bandwidth is exceeded at either site.
- If the SIP tie-trunk is unavailable, the ShoreTel system will not failover and route the call through the parent site when the following are true:
  - — The site is connected to the parent site by an emergency-enabled SIP tie-trunk.
  - — Parent as proxy is enabled for the site.
  - — The site has no available emergency-enabled trunk.

NOTE At sites with multiple trunks, the trunk selection order is SIP, ISDN, Digital, Analog. Additionally, when trunk groups are configured in the ShoreTel system, the default programming enables emergency services in each trunk group.

System administrators should consider that emergency calls will be routed over SIP if a SIP trunk is available and are encouraged to configure a dedicated, non-SIP, non-emergency trunk and disable Emergency Services in SIP Trunk Groups.

IMPORTANT A dedicated emergency-enabled trunk must be configured at each site to ensure emergency calls always reach the CO and PSAP.

Call permissions are ignored when an emergency call is placed to ensure that a user can dial emergency number from any extension on the system, regardless of the permissions associated with that user or the extension from which he or she is calling.

Once the user dials an emergency number, the call leaves the extension, arrives at the switch, and is routed to any available emergency-capable trunk at the originating site. If the user belongs to a user group that does not have access to any emergency-capable trunks, then the call will not be placed.

WARNING When adding users to the ShoreTel system, make sure each user is placed in a user group that has access to an emergency-capable trunk group. If a user is placed in a user group that does not have access to an emergency-capable trunk (e.g. a user group with long distance trunks only), members of that user group will not be able to dial emergency numbers, and they will get a reorder tone when attempting to do so.

To better understand this, you must realize that users are placed into user groups when added to the ShoreTel system. The user groups are assigned to trunk groups, and these trunk groups have different capabilities, one of which is the ability to place emergency calls. If a user belongs to only one user group, that group must have access to an emergency-capable trunk. It is crucial that each site have at least one emergency-capable trunk.

For details on adding users to a user group that has access to an emergency-capable trunk, see "User Groups" on page 325.

Always confirm with your service provider that a trunk supports emergency calls. In some instances, this may not be the case (for example, with long-distance trunks). If the trunk does not support emergency, be sure to un-check the emergency parameter as an available service in the associated trunk group in Director.

If you have mistakenly set up a site that has no available emergency-capable trunks, emergency calls will be routed to the emergency-capable trunk at the proxy site if one has been designated. By routing the call to a proxy site, the ShoreTel system is making a “last ditch” attempt to place the emergency call. This failover behavior can be unreliable and should not be relied upon to ensure that users on your system can dial emergency numbers. If you use the “parent as proxy” configuration, make sure the boundary between the two sites never traverses geographic locations that would send an emergency call to the incorrect emergency-service provider. For example, if improperly configured, a caller in Houston could pick up a phone, dial 911, and reach a 911 service in Boston because the system was configured to have the Boston site as the parent of the Houston site with “parent as proxy” checked.

Each site should have at least one emergency-capable trunk. If there will only be one trunk at a particular site, that trunk should be capable of placing an emergency call. You should also be aware that if there is only one trunk at a site, only one emergency call can be placed at a time. Therefore, you should make sure you have enough emergency trunks at each site to accommodate the (realistic) potential emergency traffic for that site.

WARNING If ShoreTel VPN phones are to be deployed in locations that are different from the site with which they are associated, placing an emergency call from a ShoreTel VPN phone needs special consideration.

In the default case, an emergency call dialed from a VPN phone will be sent to the PSAP associated with the site that hosts the switch and VPN concentrator. The emergency call would be answered but likely by a response center that is out of area for the VPN phone user which could delay or prevent an appropriate response.

ShoreTel strongly recommends that you deploy a 3rd party solution that can send a VPN phone's emergency call to the appropriate response center. Otherwise you should clearly mark VPN phones to alert users that emergency calls should not be attempted from such phones and you should educate your VPN phone users about the emergency-number limitations of the VPN phone.

A.3.3 Trunk Signaling for Emergency Calls

When an emergency call is routed out an analog or digital loop-start or a digital wink start trunk, the service provider is responsible for passing caller ID information to the PSAP.

When an emergency call is routed through a T1 PRI trunk, the ShoreTel System sends the proper caller ID information to the service provider, and the service provider must forward the information to the PSAP.

Contact your local telecommunications service provider to communicate your emergency implementation plans and have them approved. It is important to ensure that the service provider will accept, and subsequently pass to the PSAP, the caller ID information configured within the ShoreTel system. In some cases, without proper planning, a provider will reject the caller ID information as configured in the ShoreTel system and will simply pass the caller ID information associated with the trunk to the PSAP. If this happens, the dispatcher may get a number telling them to go to the wrong location.

User's have a "home port" defined in ShoreTel Director. If a user is not at his home port, it could change the caller ID number delivered to the service provider on emergency calls.

For mobile workers who travel between sites, the user must have access to an emergency-capable trunk at every site. In remote locations, the user should use the emergency trunk associated with that remote location.

A.4 Selecting Caller ID Type for Emergency Calls

There are a number of different caller ID choices available within ShoreTel that can be used by the PSAP to identify callers when they place an emergency call. The list below summarizes the available choices for sending the caller ID to the service provider for emergency calls. Options are listed in the order of precedence, meaning that if the first item on this list is not configured within the ShoreTel system, then the next piece of information on the list will be sent. Additional details about each of these caller ID options appears after the list.

1. User's Caller ID number
2. User's Direct Inward Dialing (DID) number
3. Caller's Emergency Service Identification ID (CESID) for an IP address range

   The CESID is the telephone extension that a switch sends to a Public Safety Answering Point (PSAP). A CESID helps to locate callers who require emergency services.
4. CESID of the controlling switch
5. CESID of the site
6. Nothing sent by ShoreTel system (the service provider sends the caller ID number associated with the trunk)

For details on selecting the best choice for your situation, refer to "Available Caller ID Options" on page 601.

If you are configuring a system in the Netherlands, please see "Special Considerations for Netherlands" on page 611.

A.4.1 Available Caller ID Options

User's Caller ID number – Each user can be assigned a caller ID number that will identify him during outbound calls. This caller ID number is typically used for outbound calls from the ShoreTel system when you do not want the receiving party to know the calling party's DID number. For example, an ACD agent may use caller ID to ensure that returned calls will go to a queue of sales agents, rather than directly to his desk. Similarly, this caller ID number can be sent to the service provider to identify the user when he places an outbound emergency call. The user's caller ID number is a very specific way of identifying the location of an individual user and is therefore likely to become less accurate over time as the PSAP's emergency database becomes out of date. Sending the CESID for outbound emergency calls is best for smaller organizations (see Figure A-2) and is defined on the User Edit page. You must select the "Send Caller ID as Caller's Emergency Service Identification" check box on the user group page.

In the scenarios described above and below, the user's caller ID number will only be sent when the user is at his home port. If the user is not at his home port, then the next available caller ID type is sent.

User's DID number – The DID number (Direct Inward Dialing) is the number someone dials from outside the ShoreTel system to reach a user at her desk. The DID is what most people would consider to be a "normal" telephone number. This DID number can be sent to the service provider to identify the user when she places an outbound emergency call. Although this is the most granular way of identifying users, it is also the most likely to become out of date in the PSAP's emergency database as people come and go. Sending the DID number for outbound emergency calls is most appropriate for smaller organizations (see Figure A-2) and is defined on the User Edit page. You must select the "Send DID as Caller's Emergency Service Identification" check box on the user group page.

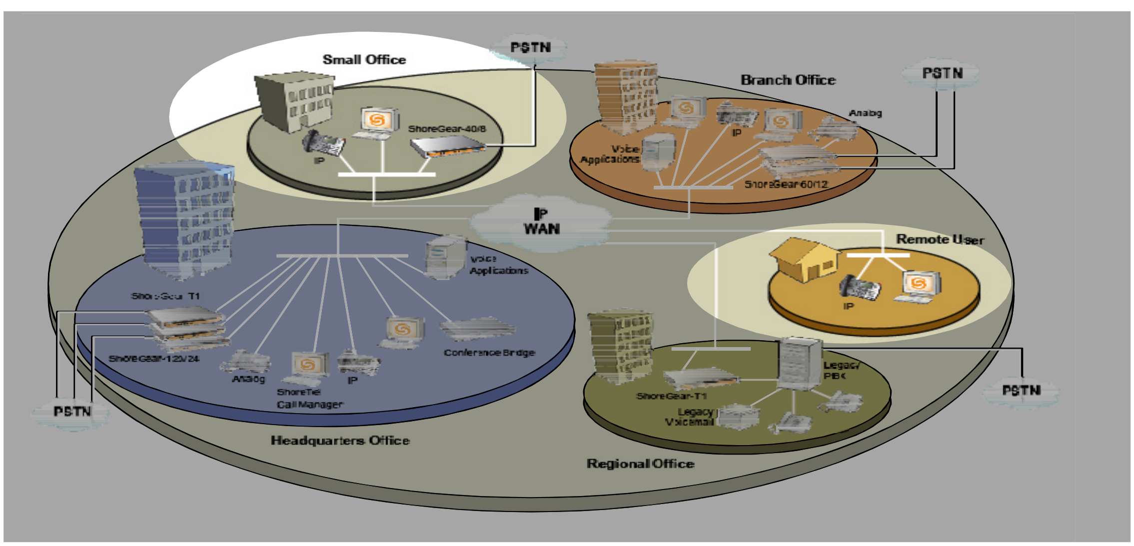


Figure A-2 Caller ID Options for Small Offices

CESID of the Specified IP Address Range – The CESID of an IP address range can also be delivered to the service provider during outbound emergency calls. A single CESID number is assigned to a range of IP addresses such that any IP phone that has an IP address that falls within the specified range will have this CESID sent for outbound emergency calls.

This option works best for identifying a phone in an office that has many floors and many extensions. Typically, a specific IP address range is configured for each floor of a building so that all users on that floor use the same CESID for emergency calls.

If a DHCP server is present, an IP phone will automatically receive an IP address within the specified range when it is connected to the network.

Sending the CESID for a specified IP address range for outbound emergency calls works best for larger organizations where simply identifying the site's street address would not provide enough information for an emergency response team to locate the caller (see Figure A-3). Furthermore, this option offers the best flexibility, the highest accuracy, and is the least likely to become out of date in the PSAP's emergency database. This option is defined on the IP Phone Address Map page.

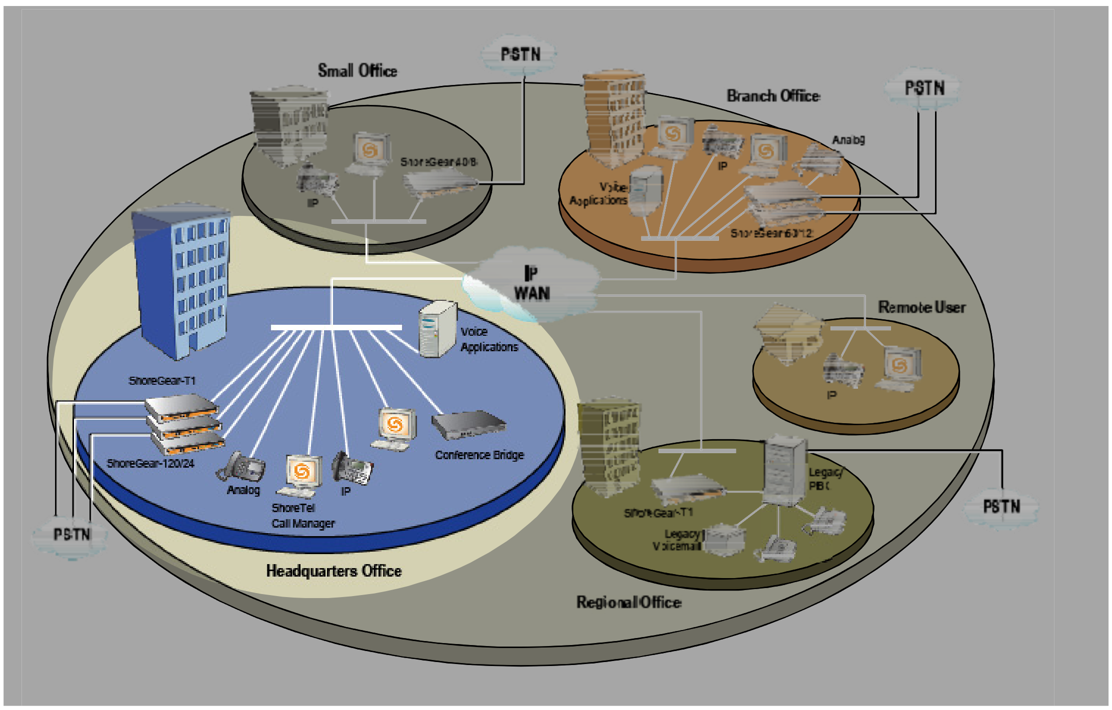


Figure A-3 IP Address Mapping – Best for Larger Offices

CESID of the controlling switch – Similar to the previous option, the Caller's Emergency Service Identification ID (CESID) of the controlling switch can also be sent to the service provider during outbound emergency calls. With this option, a CESID number is assigned to a phone switch and for any phone plugged into this switch, the switch's CESID is sent for outbound emergency calls. This option is best for larger organizations in which users are calling from analog phones (see Figure A-3). Using the IP Phone Address Map method will not work with analog phones. This approach ensures that the emergency response team is sent to the approximate vicinity of the calling party. This option is defined on the Switch Edit page.

Site (Caller's Emergency Service Identification (CESID) – This option delivers the CESID associated with the site to the service provider during emergency calls. This approach might not be granular enough for larger enterprises, but it could work well for single-site organizations or for situations in which it would be adequate to provide the emergency response personnel with a building address. This option is defined on the Site Edit page. (See Figure A-4).

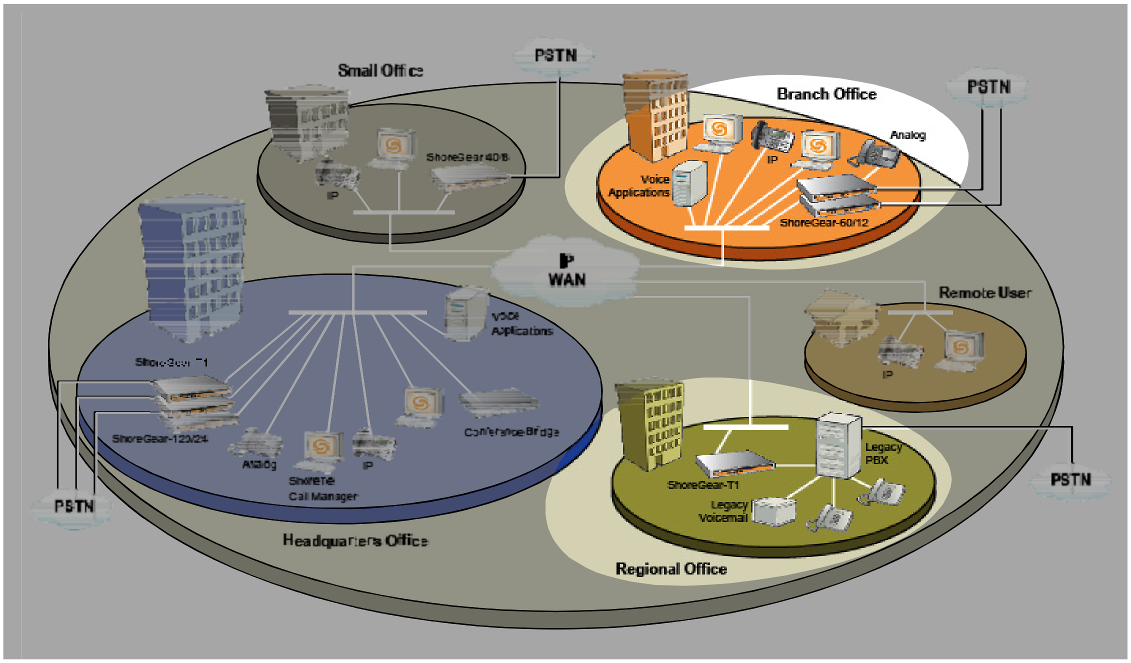


Figure A-4 Send the Site CESID for Medium-Sized Offices

Table A-1 shows several common customer scenarios and provides recommendations for how E911 should be configured, along with reasons why that is the preferred configuration.

Rules and regulations for E911 vary from one region to the next.Check with the local public safety agency to ensure you configure the system to meet the necessary requirements.

Table A-1 E911 configuration options

Scenario	Note
Small site with analog trunks	• No emergency configuration necessary
College dormitory rooms (with PRI)	• Emergency response personnel must be dispatched to a specific room • Consider sending Caller ID or DID (we recommend turning off extension assignment for this application)
Classroom (with PRI)	• Emergency response personnel must be dispatched to a specific room • Send DID or Caller ID (Consider turning off extension assignment)

Table A-1 E911 configuration options (Continued)

Scenario	Note
Multi-building campus or office complex (with centralized PRI)	• Caller ID or DID may be too granular with too much management overhead. • Consider using IP phone address mapping and/or the switch CESID.
Large building with multiple floors (with PRI)	• Caller ID or DID may be too complex. • Consider using IP phone address mapping and/or the switch CESID.
SoftPhones or travelling user	• Use home phone or hotel phone for emergency calls.
Remote IP phones (with PRI at headquarters)	• Dial an emergency number with home phone. • Use the IP phone address map (home CESID) as a backup.
VPN Phone – Fixed Location	• Remote worker install phone once, then never moves it. • Configure Caller ID of phone to reflect geographic location. One option is setting Caller ID to be identical to worker's home phone number.
VPN Phone – Variable Location	• Remote worker uses phone when traveling from various locations. • Use home phone or hotel phone for emergency calls.

A.5 Configuring Your System

The following subsections provide information for configuring your ShoreTel system for proper emergency operation.

A.5.1 Trunk Groups

Make sure you have an outbound trunk group with outbound access that also supports the emergency trunk service. If there is no emergency-capable trunk group configured, you should create one on the appropriate Trunk Group edit page (see Figure A-5).

To configure a trunk group to support emergency service, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Users > Trunks > Trunk Groups**. The Trunk Groups page appears.

Step 3 Select the trunk that you want to configure to support emergency dialing. The Edit Trunk Group page appears as shown in Figure A-5.

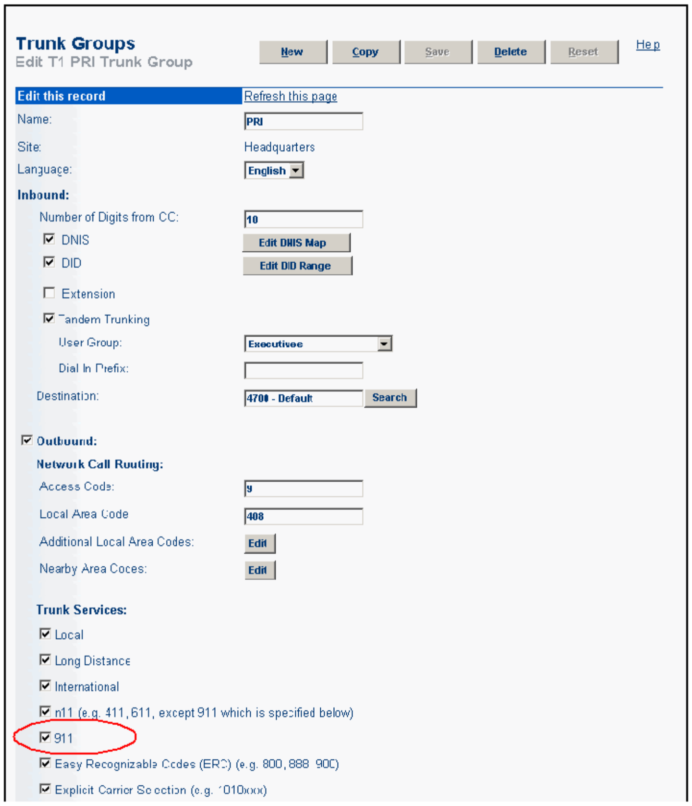


Figure A-5 Trunk Group Edit Page (T1 PRI Trunk Group)

Step 4 Check the **Emergency (e.g. 911)** check box.

Step 5 Click **Save**.

As a precaution, you should review all other trunk groups to make sure that the Emergency check box is not inadvertently enabled on a trunk that is not emergency-capable.

A.5.2 User Groups

Make sure each user group has access to a emergency-capable trunk group. You can select the desired emergency Caller ID choice on the **User Group** edit page.

- To send the Caller ID as the CESID number, verify the **Send Caller ID as Caller's Emergency** check box is selected.
- To send the DID as the CESID number, verify the **Send DID as Caller's Emergency** check box is selected.

To enable a user group to support emergency dialing, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Users > User Groups**. The User Groups page appears as shown in Figure A-6.

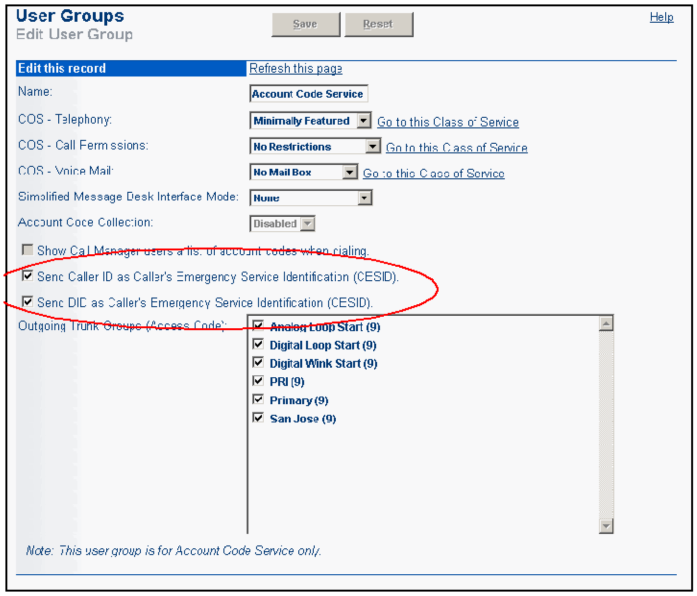


Figure A-6 User Group Edit Page

Step 3 Check the **Send Caller ID as Caller's Emergency** check box to send the Caller ID as the CESID number.

Step 4 Check the **Send DID as Caller's Emergency** check box to send the DID as the CESID number.

Step 5 Click **Save**.

Make sure you give access to trunk groups at other sites in case users in the group use the Extension Assignment feature from another site.

A.5.3 Users

Make sure the **Caller ID** field is configured if you are sending Caller ID as CESID for this user. Similarly, make sure the **DID** check box is selected (and contains a valid number in the **DID** field) if you are sending DID as CESID for this user.

Verify each user belongs to the correct user group. Use the Edit User page (Figure A-7) to associate users with the appropriate user group. See "Individual Users" on page 329 for more configuration information.

To configure a user to send Caller ID as CESID, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Users > Individual Users**. The Individual Users page appears.

Step 3 Select the user that you want to send caller ID as CESID. The Edit User page appears as shown in Figure A-7.

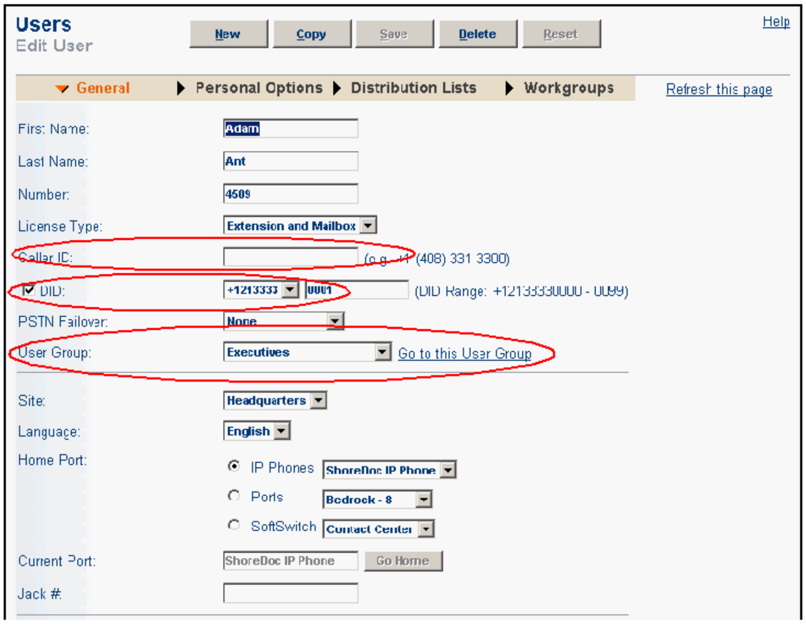


Figure A-7 Edit User Page

Step 4 Do one of the following:

- In the Caller ID field, enter the number that you want to send for this user.
- Check the **DID Range** check box and make sure there is a valid number listed in the field.
- In the DID field, enter the DID number that you want to use for the user.

Step 5 In the User Group field, select a user group that has the type of emergency support enable that you want the user to use.

Step 6 Click **Save**.

You cannot configure any user (including workgroups or route points) with a 911, 911x, or 911xx extension. These extension ranges are reserved for the 911 feature operation. We recommend that you take precaution when creating numbers that they do not conflict with the local emergency dial number.

A.5.4 Specifying CESID for IP Phone Address Range

When you have sites in different geographical areas, you must make sure that the correct local emergency number is associated with the site. You can do this by associating the local CESID number with the IP address range the system uses to assign number to new phones at the site. To associate the CESID with numbers assigned to phone at the site, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > IP Phones > IP Phones Address Map**. The IP Address Map List page appears.

Step 3 Click the **site** to which you want to associate the local CESID. The IP Phone Address Map Info dialog box appears as shown in Figure A-8.

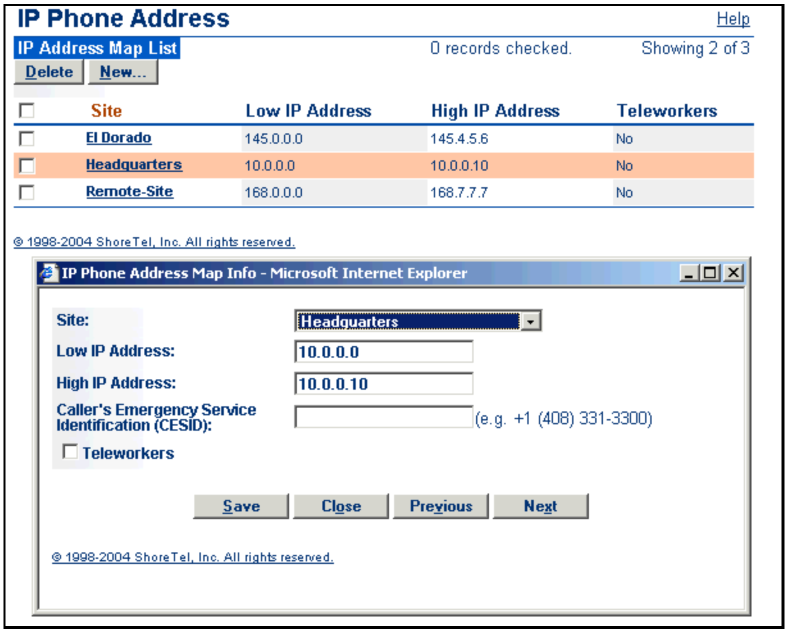


Figure A-8 IP Phone Address Map Dialog Box

Step 4 In the Caller Emergency Service Identification (CESID) field, enter the emergency phone number that is local for the site.

Step 5 Click **Save**.

A.5.5 Switch

To configure a switch with a CESID, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Platform Hardware > Voice Switches/Service Appliances > Primary**. The Primary Voice Switches/Service Appliances page appears.

Step 3 Select the switch that you want to configure with **a CESID number**. The Edit Switch page appears.

Step 4 In the Caller's Emergency Service Identification (CESID) field, enter the CESID number that is local for the area the switch services.

Step 5 Click **Save**.

A.5.6 Sites

Use the **Site** edit page to configure a site's CESID number. See Chapter 3: ShoreTel Sites starting on page 65, for additional information about configuring sites.

To configure a site to support emergency numbers, do the following:

Step 1 Launch ShoreTel Director.

Step 2 Click **Administration > Sites**. The Sites page appears.

Step 3 Select the site that you want to configure to support emergency numbers. The Edit Site page appears as shown in Figure A-9.

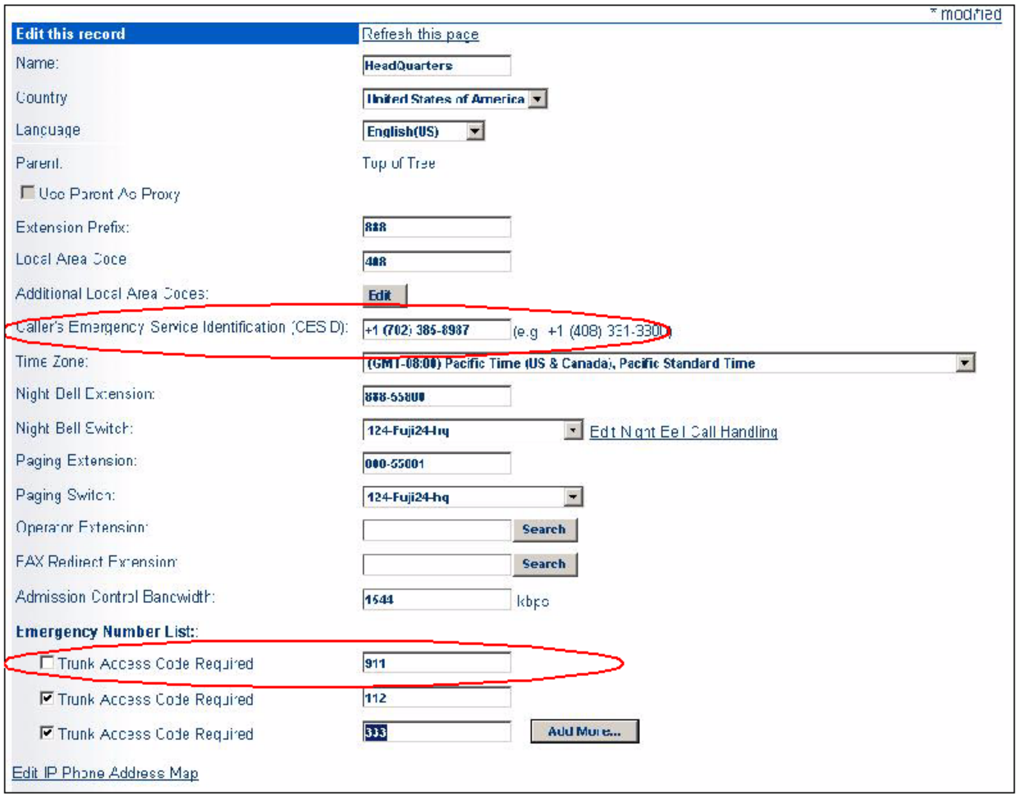


Figure A-9 **Site Edit Page**

Step 4 In the Caller's Emergency Service Identification (CESID) field, enter the number that you want the site to send for emergency responses.

> **NOTE** Make sure this field is configured with the appropriate number for the country or area the site services. For example, sites serving phones in the United States and Canada use 911.

Step 5 In the Emergency Number List section, check the **Trunk Access Code Required** check boxes with the trunk access code over which you want to send emergency calls.

Step 6 Click **Save**.

A.6 Planning Your Emergency Response

When an emergency call is made, the system automatically generates an event in the Windows event log at the beginning of the call. With the use of an event filter, you can automatically send an e-mail message to the appropriate people in your organization to help coordinate your local response (i.e. at the organizational level) whenever the emergency number is dialed.

We recommend training the personnel at all sites on the emergency operations of your ShoreTel IP voice system. All users should know how to access emergency services during normal and power outage situations.

A.6.1 Call Notification

You can set up an event filter to generate an e-mail message to help coordinate your emergency response. For more information about event filters, see "Event Filters" on page 563 for more configuration information. Use the Event Filter edit page to configure the event filter for the following parameters:

Step 1 Launch ShoreTel Director.

Step 2 Click **Maintenance > Event Filter**. The Event Filters page appears.

Step 3 Click **Add New** to create an event filter for emergency calls. The Edit Event Filter page appears as shown in Figure A-10.

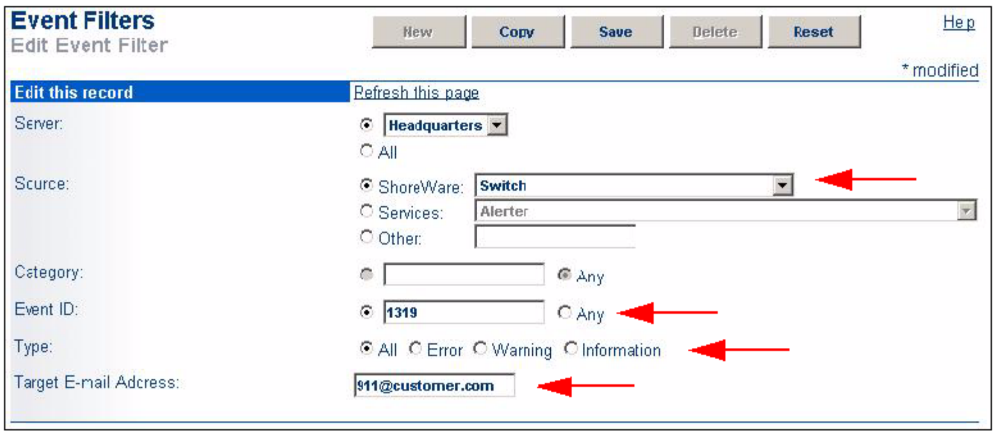


Figure A-10 Event Filter Edit Page

Step 4 In the Server section, select the server that you want to monitor for emergency events.

Step 5 In the Source section, select switch.

Step 6 In the Event ID field, enter 1319.

Step 7 In the Type section, select All.

Step 8 In the Target email address field, enter the email address of the party to whom you want emergency notification sent.

Step 9 Click **Save**.

Figure A-11 shows a typical logging message that would result after an emergency call was placed – assuming notifications had been properly configured.

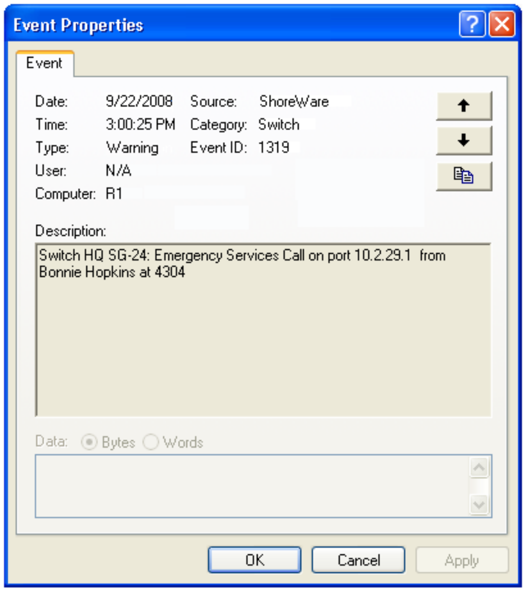


Figure A-11 Event Log

We suggest naming your switches with location information such that you can understand which site the call was made from.

A.7 International Emergency Numbers

The ShoreTel system allows dialing of emergency numbers with and without trunk access codes. For this reason, you should reserve the dialing plan space for this feature. For example:

- “112” is used in Europe and other countries.
- “000” is used in Australia.
- “999” is used in Asia.

Thus, to make use of this feature, extensions should not begin with “112”, “911,” or “999”.

NOTE Extensions should never begin with “0”.

Each site can have a maximum of ten emergency numbers to accommodate locations where multiple emergency service numbers are required.

For more information on international installations, see the ***ShoreTel 12: Planning and Installation Guide***.

A.7.1 Special Considerations for Netherlands

It is against the law in the Netherlands to “spoof” Caller ID. Caller ID will only be sent if the configured caller ID corresponds to the incoming DID that is associated with a particular trunk.

Any number entered in the CESID field in the “Switch Edit Page” and “Site Edit Page” will only be sent if the number matches the number associated with the incoming DID for that trunk.

A.8 Verifying Your Emergency Configuration

After you have finished configuring your system for emergency operation, we recommend working with your local emergency dispatch center to test your configuration in order to verify that it has been correctly configured, is sending out the desired caller ID information, and is dispatching emergency response personnel to the proper location.

We recommend calling your local law enforcement agency's non-emergency number to understand how to go about the test and to arrange a call time during non-peak hours. Do not place your emergency test call without making prior arrangements! Depending on your location, an officer may be required on-site when making test calls.

Table A-2 is intended to help you plan your test call to the local dispatch center.

Table A-2 Emergency Call Test Matrix

Site	Extension	User	Expected Caller ID	Actual Caller ID	Pass or Fail

A.9 Additional Recommendations

All sites should be configured with a designated power failure emergency phone configured appropriately. Each designated power failure emergency phone should be configured on the following ports, based on type of switch, to take advantage of ShoreTel's emergency line power failure feature:

- SG40 – Port 4: Analog Trunk; Port 5: Analog Emergency Phone
- SG60 – Port 8: Analog Trunk; Port 9: Analog Emergency Phone
- SG120 – Port 8: Analog Trunk; Port 9: Analog Emergency Phone
- SG30 – Port 1: Analog Trunk; Port 12: Analog Emergency Phone
- SG50 – Port 1: Analog Trunk; Port 12: Analog Emergency Phone
- SG50V – Port 1: Analog Trunk; Port 12: Analog Emergency Phone
- SG90 – Port 1: Analog Trunk; Port 12: Analog Emergency Phone
- SG90V – Port 1: Analog Trunk; Port 12: Analog Emergency Phone
- SG220T1A – Port 1: Analog Trunk; Port 12: Analog Emergency Phone

APPENDIX *B*

Call Detail Record Reports

Call detail record (CDR) reports are used to review the ongoing call activity on the ShoreTel system. Sections in this appendix include:

B.1 Overview

The ShoreTel system tracks all call activity and generates call detail records into a database as well as into a text file on the ShoreTel server. The call detail records are used to generate CDR reports. The system comes bundled with twelve CDR reports based on data from the CDR database. In addition, the text files provide third-party call accounting systems with a simple and standard way to access the call data.

If the ShoreTel server is not up and running, call detail records will not be generated, and the associated calls cannot be presented in the CDR reports.

In the event of a WAN outage, CDR data is stored for up to two hours on the distributed server. When WAN connectivity is restored, the stored data is forwarded to the Headquarters database. After two hours, the distributed server deletes the data and logs an error to the NT event log.

B.1.1 Call Accounting Service

The ShoreTel system has a call accounting service running on the main server that generates call detail records into a database and into a space-delimited text file for use by third-party call accounting applications. The call accounting service is also responsible for archiving all the CDR data. The CDR files are located under C:\Shoreline Data\Call Records 2.

B.1.2 Active CDR Database

The call accounting service generates call detail records into the active CDR database. This file includes all call activity for the period of time specified in the Retention Period for CDR Data parameter in the Director Reporting Options page, as shown in Figure B-1. To access this page, select Reporting > Options from the Director menu.

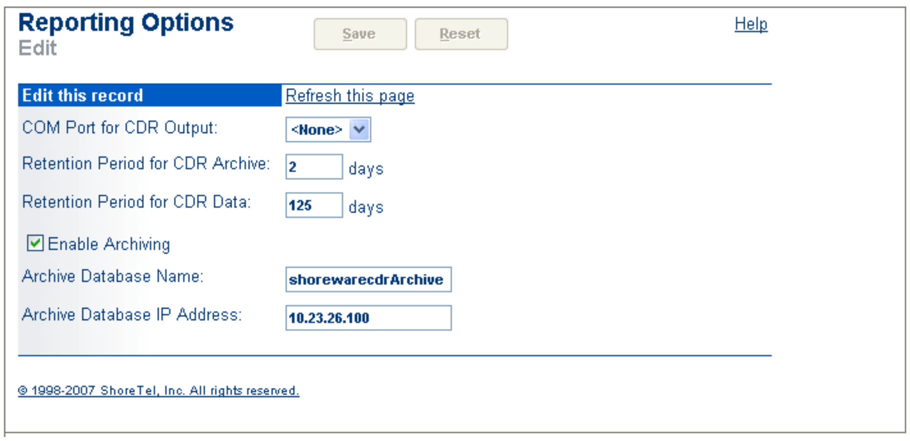


Figure B-1 Reporting Options Page

When Enable Archiving is selected on the Reporting Options page, a nightly routine automatically moves call detail records that are older than the limit specified by the Retention Period for CDR Data into the Archive database.

B.1.3 Legacy CDR Text Files

The call accounting service automatically generates a daily legacy CDR text file for use by third-party call accounting applications. These packages typically provide numerous reports, including:

- Call accounting, cost allocation
- Most frequently dialed numbers
- Most costly dialed numbers
- Most costly users
- Trunk utilization
- Toll fraud

The CDR*.log files are text files created daily at midnight. It contains call records from midnight to midnight. Any call records that span the midnight hour will be recorded on the day that calls are completed.

B.1.3.1 Format

The file name format for the daily **CDR-YYMMDD.HHMMSS.log** where

- YY, MM, and DD are zero-padded character strings that represent the year, month, and day of the date when the file was created.
- HH, MM, and SS are zero-padded character strings that represent the hour, minute, and second of the time when the file was created.

Call records are entered in the log file in the order of when the call was completed and not when it began.

It is the responsibility of the third-party reporting application to delete the daily log files.

The format of the record is column based, must be justified correctly, and end with a carriage return and line feed. A single blank character is inserted between each data field for readability. Table B-1 provides information about elements in the CDR text file.

Table B-1 CDR Text File Field Definitions

Field	Column	Length	Comment
Call ID	1	10 right justified	A unique ID that represents the call. The Call ID is meant to be unique for the duration while it's active.
Date	12	10	Date of the call given in month, day, and year: mm/dd/yyyy
Time	23	8	Start of the call given in hours, minutes, and seconds: hh:mm:ss
Extension	32	16 left justified	Inbound or outbound extension ID. Last valid party on the call. Valid parties include user extension and operator but not voice mail or auto-attendant.
Duration	49	8	Call duration given in hours, minutes, and seconds. hh:mm:ss
Call Direction	58	1	Incoming/outgoing flag 0 – Incoming 1 – Outgoing 2 – Tandem Trunking – Inbound Tandem Call
Dialed Number	60	16 left justified	Contains the number dialed but does not include any access code, such as 9, to seize the trunk. Valid only for outbound calls.
Caller ID	77	16 left justified	Blocked or unavailable information will be reported as blocked or unavailable in text. Valid only for incoming calls.
Trunk Member	94	4 right justified	The Port ID of the trunk.
Trunk Group	99	3 right justified	The Trunk group ID.
Account Code	103	20	The account code entered by the caller.
CR /LF	124		Carriage return.

B.2 Talk Time Record

Call Detail Records provide a complete logging of all non "Private" calls placed in or out of the ShoreTel system. These logs include call volume, call origination, call destination and the length of each call.

The Talk Time Enhancement feature increases the usability of log data from the Call Detail Records by compiling only the actual time spent in a conversation between the calling parties. All call ring back time is eliminated from the Call Detail Records retaining only the actual Talk Time spent on the call.

When a call is placed the destination phone acknowledges and a ring tone is provided to both parties. The time of the call starts on the first ring and terminates when the calling parties hang up. The ShoreTel Appliance uses the Telephony Management Service (TMS) to report the call and it's the time to the HQ server. There the call is captured in the Call Detail Records. The entire length of the call is logged here, including the first ring up to the entire call tear down.

Talk Time Enhancements uses the FarEndAnswered event provided by certain trunks to determine when the called party answers the call. The length of time the call has been answered is reported to the CDR.

Calls placed over Digital Wink, PRI, BRI or SIP trunks support FarEndAnswered events. These events mark the moment when the called party answers the call. Talk Time Enhancement uses this event to report the Talk Time of the call to the Call Detail Records. The Call time represents only the actual Talk Time of the call. Calls over Analog loop start and Digital loop start trunks do not support FarEndAnswered events. These calls still report to the CDR through TMS, but their time values include the time for Ring Back.

As administrators evaluate CDR reports, an understanding of the trunk types that support Talk Time Enhancement helps them to gain an accurate picture of the talk time being reported in the CDR.

B.3 MySQL Database

CDR records are maintained and queried through MySQL database. The maximum MySQL database size is 64 terabytes (TB). Database tables restricted to a maximum size of 2 TB.

The data in the MySQL files can be viewed using a new Web-Based Reporting feature from ShoreTel Director (see the "Accessing the Call Details Reports Page" on page 587 for more information). Alternatively, the user can use common database command utilities via a command line interface to dump and restore files.

CDR reports are generated from Director, as described in "Accessing the Call Details Reports Page" on page 587.

MySQL database service can be monitored or restarted from Director by selecting Maintenance > Services in the menu page, then selecting MySQL in the table on the Services page.

B.3.1 Compatibility and Pre-Configuration Requirements

B.3.1.1 Disk Space Requirements

Storing call detail records for 50,000 workgroup calls requires a 1.5 GB MySQL database. Implementing a database of this size typically requires 4.0 GB of disk space. This includes disk space for the main database (1.5 GB), the archive database (1.5 GB), and temporary space required to generate reports (1.0 GB).

Although the main and archive databases are typically stored on the same server, MySQL permits the storage of the databases on different servers.

B.3.1.2 Compatibility with Utility Programs

ShoreTel should be run on a dedicated server. Other programs that access MySQL databases may not be compatible with ShoreTel, resulting in installation and data integrity issues. Before installing ShoreTel on any server, remove all pre-existing MySQL programs and databases.

Virus Checkers: MySQL database files must be excluded from all virus checker utilities running on the server. Specifically, if virus checker is running on the server, exclude MySQL CDR Database file (Where ever ShoreLine Data installed:\Shoreline Data\Call Records 2\Data\[ibdata1, ib_logfile0, ib_logfile1]) from your virus checker utility. If these files are not in the exclusion list, MySQL service will stop working.

Disk or Backup utilities: MySQL database files must be excluded from all disk or backup utilities running on the server. Failure to exclude the database will crash the MySQL service.

To restart the database after a crash, access the MySQL Service page from ShoreTel Director by selecting Maintenance > Services in the menu page, then selecting ShoreTel MySQL in the table on the **Services** page.

B.3.2 Archival and Backup Utilities

The following section describes database archival, backup, and replication tools. Table B-2 summarizes the service availability for these features.

Table B-2 Archival, Backup, and Replication Services Availability

Field	Backup	Archive (Secondary Server)	Replication
Technical Assistance Support	Yes	Yes	No
Additional MySQL License Required	No	Yes	Yes
Execution	Manual	Daily	Online
Reports run on remote machine	No	Yes	No
Complete restoration if HQ fails	Yes	possible through manual recovery	Yes

B.3.2.1 Record Retention Periods

You configure the number of days that a database stores a Call Detail Record in the Report Options page. To access the Report Options page, as shown in Figure B-2, open ShoreTel Director and select Reporting > Options from the Director Menu.

- **Retention Period for CDR Data** specifies the number of days that records remain in the main CDR database. Older records are removed from the database each day.
- **Retention Period for CDR Archive** specifies the number of days that records remain in the archive database. Older records are removed from the database each day.

Default values for each parameter is 125 days.

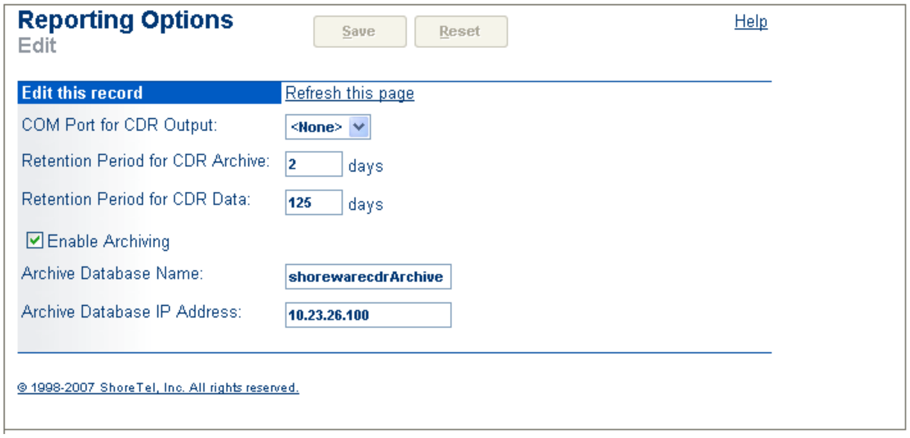


Figure B-2 Report Options Page

B.3.2.2 Database Archive Utility

The archive utility provides a method of removing older records from the main database and storing them in an archive database. Archiving older records into a separate database reduces the storage requirements of the main database, which reduces the time required to search for specific records or generate reports. The archived database provides the same set of services as the main database.

Archival services are configured and enabled in ShoreTel Director, where you can specify the number of days that records are maintained in the main database and in the archive database. When archiving is enabled, archival services are performed daily. The archival service copies records to the archive database that exceed the main database age limit, then removes those records from the main database. Records that exceed the age limit for the archive database are removed from the archive database. Age limits are established separately for each database; valid limits range from one to 2000 days. The default age limits for each database is 125 days.

Example: A sample implementation sets a 30 day limit on the main database and a 365 day limit on the archive database. In this case, the main database contains records for calls handled during the past 30 days while the archive database contains records for calls handled during the past 365 days.

The Backup utility can be used for record storage requirements that exceed 2000 days.

To create an archive database:

Step 1 Run MakeCDRArchive –d databasename, where databasename is the name of the archive database to be created.

Step 2 Access the Reporting Options page in ShoreTel Director (Reporting | Options from the Menu page) to configure ShoreTel to access the archive database.

B.3.2.3 Database Backup Utility

The Backup utility creates a copy of a specified database, which can be restored at a later time and different location. The Backup utility differs from the Archival utility as follows:

- Archiving is configured once then performed daily. Backups are performed only when a command is executed.

- Archival operations are configured from ShoreTel Director. Backups are performed from the command line.
- Archive databases can be accessed directly to generate reports. Backup databases must be restored before performing search and report generation tasks.

Backup and Restore operations can be performed without shutting down the MySQL service. Performing these operations during off peak hours reduces the execution time and the impact on other system services.

The file located at "C:\Program Files\Shoreline Communications\Shoreware Server\MySQL\MySQL Server 5.0\Examples\dump1.bat" is an example of a batch file that backs up a MySQL CDR database under generic default conditions. This file can be used as a template for creating a batch file that backs up the database under specific conditions. The password is shorewaredba. Backing up a 1.5 GB database requires 200 seconds.

Refer to http://dev.mysql.com/doc/refman/5.0/en/disaster-prevention.html for MySQL backup tools, add-ons, and documentation.

B.3.2.4 Database Restore Utility

Restoring a database copies the records in the backup database file to the database specified in the restore command. Records in the backup file that are duplicates of records in the target database are listed in the log file and are not restored.

The file located at "C:\Program Files\Shoreline Communications\Shoreware Server\MySQL\MySQL Server 5.0\Examples\restore1.bat" is an example of a batch file that restores a MySQL CDR database under generic default conditions. This file can be used as a template for creating a batch file that restores the database under specific conditions. The password is shorewaredba. Restoring a 1.5 GB database requires 1200 seconds.

B.3.2.5 Database Replication

MySQL provides a Database Replication tool. Access the following web pages for more information:

- http://www.howtoforge.com/mysql_database_replication provides MySQL database replication setup information.
- http://solutions.mysql.com/solutions/tools?type=25 lists tools and add-ons that assist with database replication.

B.3.3 Installing MySQL on a Secondary Server

Although the archive database is normally stored on the main ShoreTel server, MySQL databases can be installed on a Secondary Server to conserve main server resources. A separate licensed copy of MySQL Enterprise Server 5.0 is required to install the database on a Secondary Server.

To install MySQL on a secondary server, perform the following procedure (Replace c:\Program Files\... in the instructions to the location where MySQL is installed on the server).

Step 1 Install MySQL Enterprise Server 5.0 on a Secondary server.

Step 2 Select all the default values during installation except for the following items:

- Use root for UserID
- Use shorewaredba for the password,

- utf8 for the character set as part of the installation

Step 3 Backup the file c:\Program Files\MySQL\MySQL Server 5.0\my.ini from the Secondary server to a safe location (i.e. c:\MySQL_backup).

Step 4 Backup the files c:\Program Files\MySQL\MySQL Server 5.0\Data\[ib_logfile*] from the Secondary server to a safe location (i.e. c:\MySQL_backup).

Step 5 Select **Start > Administrative Tools > Services > MySQL**

Step 6 Click **Stop the service** and check that MySQL service status is blank

Step 7 Compare the file from the Main server directory with a secondary server file:

Main server directory—*C:\Program Files\Shoreline Communications\ShoreWare Server\MySQL\MySQL Server 5.0\Examples*

Secondary server file— *c:\Program Files\MySQL\MySql Server 5.0*

Make sure that all the parameters specified in the archive_MySQL_my.ini are set appropriately in my.ini.

Step 8 Archive_MySQL_my.ini values are as follows.

```
[mysql]
default-character-set=utf8
[mysqld]
default-character-set=utf8
tmp_table_size = 30M
key_buffer_size=2M
read_buffer_size=2M
read_rnd_buffer_size=2M
sort_buffer_size=2M
innodb_additional_mem_pool_size=2M
innodb_flush_log_at_trx_commit=0
innodb_file_per_table
innodb_log_buffer_size=5M
innodb_buffer_pool_size=150M
innodb_log_file_size=24M
default-storage-engine=INNODB
```

Step 9 Delete the file ib_logfile* from the Secondary server directory (c:\Program Files\MySQL\MySQL Server 5.0\Data).

Step 10 Check that innodb_flush_log_at_trx_commit=0 on Secondary server C:\Program Files\MySQL\MySql Server 5.0. If the value is not zero, archiving write operations will be more than 20 times slower.

Step 11 Select **Start > Administrative Tools > Services > MySQL**

Step 12 Click **Restart the service** and verify that MySQL service comes back up.

To convert the Secondary server database into an archive database:

Step 1 Verify the following files are placed in an equivalent location on the Secondary Server to that on the Main servers (default location is \\Shoreline Communications \Shoreware Server)

- MakeCDR.dll
- MakeCDR.sql
- MakeCDR_sp.sql
- MakeCDRArchive.exe

Step 2 Run MakeCDRArchive –d databasename, where databasename is the name of the archive database to be created.

Access the Reporting Options page in ShoreTel Director (Reporting | Options from the Menu page) to configure the name of the archive database within ShoreTel.

B.3.4 Performance Tuning for Report Generation

To improve on the CDR report generation performance, increase INNODB_BUFFER_POOL_SIZE defined in c:\windows\my.ini based as specified

Default setting:

- INNODB_BUFFER_POOL_SIZE = 150 MB

If the database contains more than 350,000 records, set

- INNODB_BUFFER_POOL_SIZE = 200 MB

If the database contains more than 500,000 records, set

- INNODB_BUFFER_POOL_SIZE = 250 MB

B.3.5 Report generation time – CPU Utilization

Based on the size of the data requested for the report, the display time for the last page of the report from the first page may require ten minutes. Even though the priority of Report Generation process is set to below normal, generating large reports may potentially impact the call processing performance. To avoid performance degradation issues, do not generate large CDR reports during peak call loads.

B.3.6 MySQL CDR Database and Internationalization

MySQL CDR Database supports the UTF-8 character set. All CDR data in the database is stored in UTF-8 character set.

B.3.7 Monitor MySQL service

To monitor and, when necessary, restart the MySQL service from ShoreTel Director, access the Services page by selecting Maintenance | Services in the menu page, then select MySQL in the table on the Services page.

B.3.8 Tools for browsing MySQL database tables

MySQL provides MySQL Query Browser as part of their GUI Tools (http://dev.mysql.com/downloads/gui-tools/5.0.html). MySQL Query Browser can be used to browse and view the queries. The open source tool (http://www.webyog.com/en/downloads.php) is available to view the CDR tables defined in MySQL.

Browsing a large CDR database on the Main server may potentially degrade the call processing server.

Large amount of temporary disk space may be used by these MySQL browser tools. To avoid affecting call processing performance on HQ, a query with LIMIT criteria can be used to show a subset of rows.

B.3.9 Restrictions in the number of records returned by the MySQL CDR query

CDR database queries that exceed 300,000 records may cause performance degradation when generating certain reports, such as Trunk Activity Detail and Trunk Activity Summary. Increasing the amount of free disk space may mitigate this problem, as will modifying the query filter to reduce the number of records returned by the query to under 300,000.

B.4 CDR Reports

The ShoreTel system comes bundled with 12 CDR reports generated using data from the active CDR database on the ShoreTel server. CDR reports present information about users, trunks, WAN links, workgroup queues, account codes, and workgroup agents. Reports are grouped into two categories: summary and detail.

Summary reports provide a high-level view of activity that occurred in a particular area, while detail reports provide a detailed view of activity beyond that of the summary report. Typically, you use the summary report to identify any discrepancies or problems, and the detail report to uncover more specific information. The workgroup queue report only has a summary report.

- **User Activity Summary:** Summarizes all calls for each user.
- **User Activity Detail:** Lists every call for each user.
- **Trunk Activity Summary:** Summarizes all calls for each trunk.
- **Trunk Activity Detail:** Lists every call for each trunk.
- **Workgroup Agent Summary:** Summarizes all inbound workgroup calls for each agent.
- **Workgroup Agent Detail:** Lists every inbound workgroup call for each agent and optionally, outbound calls. Non-workgroup calls for the agent are also reported.
- **Workgroup Queue Summary:** Summarizes queue activity for every workgroup, including calls that went directly to agents.
- **Workgroup Service Level Summary:** Summarizes data on call processing by the workgroup server.
- **WAN Media Stream Summary:** Summarizes media stream traffic and call quality for calls made over the WAN in multi-site deployments.
- **WAN Media Stream Detail:** Lists media stream made over the WAN in multi-site deployments.
- **Account Code Summary:** Summarizes call information for each account; counts of calls each day, along with their total and average duration. There are also totals for the reporting period.
- **Account Code Detail:** Provides a detailed list of calls that occurred for each account. For each call the date/time of the call, number dialed, the extension making the call

and the duration of the call is included. For each account, a summary is provided of the number of calls, along with their total and average duration.

B.4.1 TMS-CDR Media Stream Statistics

The TMS-CDR Media Stream Statistics feature offers a method of formatting and storing Call Detail Records (CDR) data on media streams and stores that formatted information into a log file on the system, making it easier for the ShoreTel system to integrate with various third-party SNMP monitoring tools, and enabling users to acquire a more accurate picture of the traffic patterns in their network. This information can be useful in performing load analysis, identifying peak traffic times, and assisting the customer in setting up competitive pricing strategies.

The system processes media statistics for all calls and formats the raw data into separate lines, with each line partitioned into several columns separated by a comma. Formatted data is then saved in a text file and is subjected to appropriate rollovers similar to the other ShoreTel server logs.

One media stream statistic record will be generated for each RTP stream on a call. Thus, a 3-way fully-meshed conference call would generate 6 records.

B.4.1.1 Formatting

Media statistics are collected and deposited line by line into a file. A delimiter separates one column from the previous one, with no delimiter prior to the first column and none after the last column. The column values will be left-justified and padded with spaces to the right. A value that exceeds the fixed-width column limit will be truncated so that it fits within the limit.

Each line will look similar to the line below:

value-1,value-2,value-3,…..,value-n

The Table B-3 summarizes the details of the individual columns.

Table B-3 CDR Media Stream Statistics Formatting

Column number	Type	Width	Description
1	Integer	20	ID of the line in decimal
2	String	20	Extension Number For anonymous calls, extension number is not available. In such cases, an empty string will be placed at this column.
3	String	16	Name of Extension or Trunk or Phone (UTF-8)

Table B-3 CDR Media Stream Statistics Formatting (Continued)

Column number	Type	Width	Description
4	Integer	2	Party type in decimal 0 Unknown 1 Station 2 Trunk 3 Virtual 4 Workgroup 5 AutoAttendant 6 VMForward 7 VMLogin 8 BackupAA 9 Anonymous Phone 10 Nightbell 11 Paging 12 Workgroup Agent 13 Unknown 14 RoutePoint 15 ACC 16 HuntGroups 17 GroupPaging
4	Integer	2	
5	String	32	SIP Call ID
6	String	16	Local IP Address (Switch, Trunk Switch, or IP Phone etc.) in dotted decimal form
7	String	16	Remote IP Address (Remote end point. Switch or Trunk or IP Phone etc.) in dotted decimal form
8	Integer	20	Local Site ID (Site ID of extension or trunk or phone that generated starts)
9	String	16	Local Site Name (UTF-8)
10	Integer	3	Code Type 1 ALAWPCMA/8000 (or G711A) 2 MULAWPCMU/8000 (or G711µ) 3 LINEARL16/16000 4 ADPCMDVI4/8000 5 G729AG729A/8000 6 G729BG729B/8000 7 LINEARWIDEBANDL16/16000 8 G722G722/8000 9 BV32BV32/16000 10 BV16BV16/8000 11 AAC_LC32000AAC_LC/32000 12 CustomCodec added by administrator
11	Integer	10	Payload size (in milliseconds)
12	Integer	2	Status code 0 – Normal 1 - Failure
13	String	12	Starting time of the collection in string HH:MM:SS.MSEC format
14	Integer	20	Duration (in seconds) of the collection in decimal.

Table B-3 CDR Media Stream Statistics Formatting (Continued)

Column number	Type	Width	Description
15	Integer	20	Number of received packets
16	Integer	20	Number of lost packets
17	Integer	20	Max jitter
18	Integer	20	Underruns
19	Integer	20	Overruns

This feature is applicable only on the main (i.e. headquarter) ShoreTel server, and is disabled by default. Enabling the TMS-CDR Media Stream Statistics feature requires making the appropriate changes to the registry settings.

WARNING Do not make any changes to the registry settings unless you are certain you know what you are doing!

Step 1 Click **Start** and select **Run**.

Step 2 Select the **regedit** application to display a window similar to the one shown in Figure B-3.

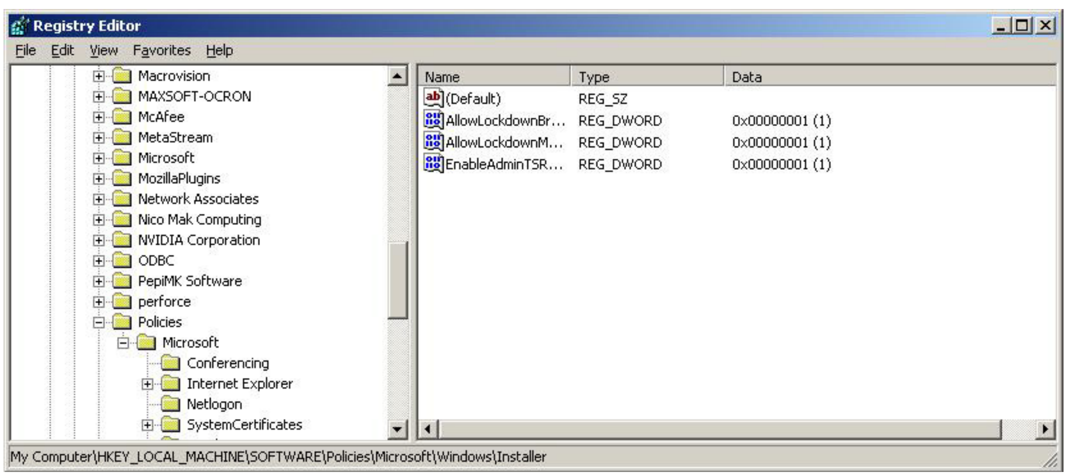


Figure B-3 Registry Editor window

Step 3 Navigate to *SOFTWARE\Shoreline Teleworks\Call Accounting*

Step 4 Double-click the file named **LogMediaStatsToFile** to open the Edit DWORD Value dialog box shown in Figure B-4.

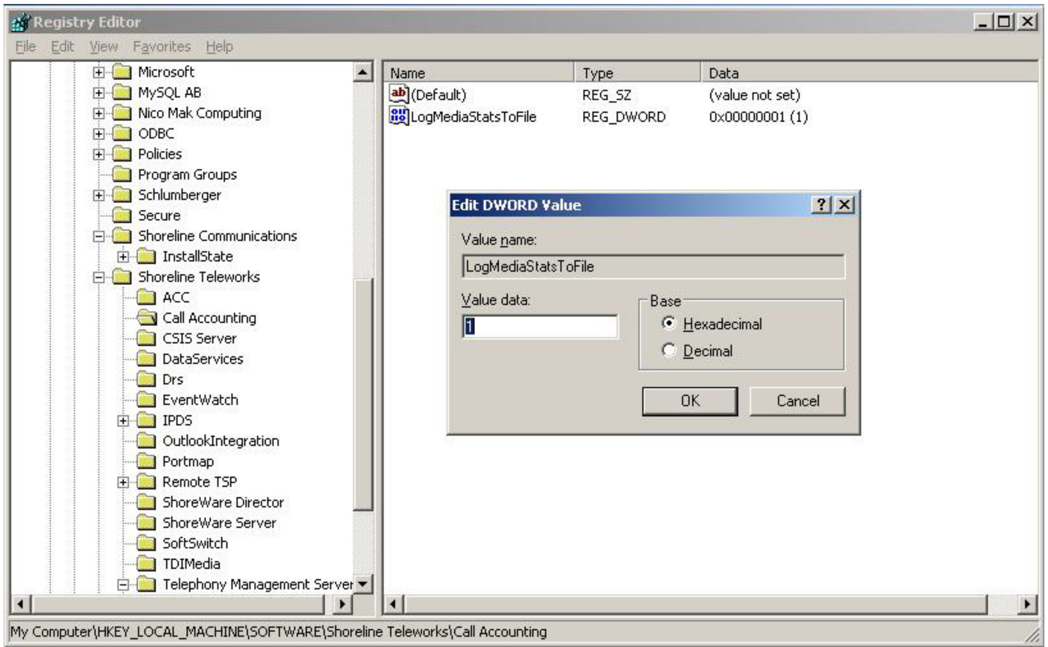


Figure B-4 Value data field

Step 5 Enter **1** in the Value data field and click **OK** to store your changes and enable this feature.

B.4.2 Generating CDR Reports

Refer to "Accessing the Call Details Reports Page" on page 587 for instructions on generating CDR Reports.

B.5 Interpreting CDR Reports

This section provides information on interpreting each of the 12 CDR reports.

B.5.1 User Activity Summary Report

The User Activity Summary Report (Figure B-5) shows a summary of all inbound and outbound calls for each user. This includes the type of calls made, as well as the total duration for all calls. This report can be run as an interval report in which user activity is subtotaled by the selected interval. Additionally, the summary can be run for selected extensions.

User Activity Summary Report

Start: | End: | Show Internal Calls: True

Start: 12:00:00AM | End: 11:59:59PM

Interval: 30 min.

	Inbound All			Outbound All			Total All			Outbound Non-Local Trun			Outbound WAN Trunk		
Name (Extension)	Qty	Duration	Average Duration	Qty	Duration	Average Duration	Qty	Duration	Average Duration	Qty	Duration	Average Duration	Qty	Duration	Average Duration
User11 (x121)															
09:00 - 09:29	0	0:00:00	0:00:00	3	0:21:00	0:07:00	3	0:21:00	0:07:00	0	0:00:00	0:00:00	0	0:00:00	0:00:00
09:30 - 09:59	2	0:02:02	0:01:01	0	0:00:00	0:00:00	2	0:02:02	0:01:01	0	0:00:00	0:00:00	0	0:00:00	0:00:00
10:00 - 10:29	2	0:04:52	0:02:26	0	0:00:00	0:00:00	2	0:04:52	0:02:26	0	0:00:00	0:00:00	0	0:00:00	0:00:00
10:30 - 10:59	3	0:09:52	0:03:17	1	0:01:04	0:01:04	4	0:10:56	0:02:44	0	0:00:00	0:00:00	0	0:00:00	0:00:00
11:00 - 11:29	0	0:00:00	0:00:00	7	0:01:59	0:00:17	7	0:01:59	0:00:17	0	0:00:00	0:00:00	0	0:00:00	0:00:00
11:30 - 11:59	1	0:00:12	0:00:12	1	0:00:06	0:00:06	2	0:00:18	0:00:09	0	0:00:00	0:00:00	0	0:00:00	0:00:00
12:30 - 12:59	2	0:01:42	0:00:51	0	0:00:00	0:00:00	2	0:01:42	0:00:51	0	0:00:00	0:00:00	0	0:00:00	0:00:00
13:00 - 13:29	2	0:00:28	0:00:14	4	0:04:59	0:01:14	6	0:05:27	0:00:54	0	0:00:00	0:00:00	0	0:00:00	0:00:00
14:00 - 14:29	3	0:09:57	0:03:19	1	0:04:10	0:04:10	4	0:14:07	0:03:31	0	0:00:00	0:00:00	0	0:00:00	0:00:00
14:30 - 14:59	1	0:00:07	0:00:07	0	0:00:00	0:00:00	1	0:00:07	0:00:07	0	0:00:00	0:00:00	0	0:00:00	0:00:00
15:00 - 15:29	2	0:00:30	0:00:15	2	0:01:03	0:00:31	4	0:01:33	0:00:23	0	0:00:00	0:00:00	0	0:00:00	0:00:00
15:30 - 15:59	0	0:00:00	0:00:00	2	0:02:12	0:01:06	2	0:02:12	0:01:06	0	0:00:00	0:00:00	0	0:00:00	0:00:00
16:30 - 16:59	1	0:05:26	0:05:26	3	0:05:26	0:01:48	4	0:10:52	0:02:43	0	0:00:00	0:00:00	0	0:00:00	0:00:00
18:00 - 18:29	0	0:00:00	0:00:00	1	0:00:14	0:00:14	1	0:00:14	0:00:14	0	0:00:00	0:00:00	0	0:00:00	0:00:00
18:30 - 18:59	3	0:10:37	0:03:32	0	0:00:00	0:00:00	3	0:10:37	0:03:32	0	0:00:00	0:00:00	0	0:00:00	0:00:00
19:00 - 19:29	4	0:15:49	0:03:57	0	0:00:00	0:00:00	4	0:15:49	0:03:57	0	0:00:00	0:00:00	0	0:00:00	0:00:00
19:30 - 19:59	4	0:03:39	0:00:54	1	0:04:50	0:04:50	5	0:08:29	0:01:41	0	0:00:00	0:00:00	1	0:04:50	0:04:50
20:00 - 20:29	0	0:00:00	0:00:00	1	0:05:11	0:05:11	1	0:05:11	0:05:11	0	0:00:00	0:00:00	1	0:05:11	0:05:11
22:00 - 22:29	0	0:00:00	0:00:00	2	0:00:02	0:00:01	2	0:00:02	0:00:01	0	0:00:00	0:00:00	0	0:00:00	0:00:00
Sub Total	30	1:05:13	0:02:10	29	0:52:16	0:01:48	59	1:57:29	0:01:59	0	0:00:00	0:00:00	2	0:10:01	0:05:00
Grand Total		**1:05:13**		**29**		**0:01:48**		**1:57:29**		**0**		**0:00:00**		**0:10:01**	
	30		**0:02:10**		**0:52:16**		**59**		**0:01:59**		**0:00:00**		**2**		**0:05:00**

Figure B-5 User Activity Summary Report

B.5.1.1 Calls Included

The User Activity Summary Report always displays External Calls and can be configured to display Internal Calls. External calls are those calls where the record in the Call table has a CallType of 2 (Inbound) or 3 (Outbound).

B.5.1.2 Name (Extension Field)

For outbound calls, the Name (Extension) field of the Call record always reports the party that initiated the call.

Inbound calls are reported according to the last party involved in the call (excluding voice mail and the auto-attendant). For example, if a call to extension 320 is not answered and the user's Call Handling Mode (CHM) forwards the call to the assistant at extension 452—who answers the call—the Extension field in the Call record contains 452.

When an inbound call is forwarded to voice mail, the Name (Extension) field records the party involved in the call before it was forwarded to voice mail. For example, if a user with extension 320 doesn't answer a call and his or her Call Handling Mode (CHM) forwards the call to voice mail, the extension field is set to 320.

The User Activity Summary Report is described in Table B-4.

Table B-4 User Activity Summary Report field descriptions

Field	Presence/ Frequency	Description
Name (Extension)	Once for each extension reported.	The name of the user, last name first. Users without a last name are presented first. Non-users such as Workgroups, Voice Mail, Voice Mail Login, and Auto-attendant are included in the report. The names for these extensions are reported for calls that only interact with these extensions (not a user extension). Like many other non-user extensions, the ShoreTel Audio Conference extension is also not displayed.
Inbound All - Qty, Duration, Average Duration	Once for each period reported.	The quantity, total duration, and average duration for inbound calls during the reporting period are presented. A call is considered inbound if the CallType field of the Call table record is set to 2 (Inbound). If the report is run with intervals, the call is only reported for the interval in which it started, even if it ends in a different interval. The StartTime field in the Call table is used to determine when the call started. Duration represents that extensions time on the call. This is found in the Connect table record's Duration field for this connection (where the Connect.CallTableID matches the Call.ID and Connect.PartyID matches Call.Extension). Since a call is reported during the period in which it starts, but may end during another interval, the duration can be longer than the 30-minute interval period—the total call duration time is reported during the interval in which the call begins. Total Duration during any period is the sum of the duration for the Inbound calls during the period. Average duration is found by dividing this total by the number of calls during the period.
Outbound All - Qty, Duration, Average Duration	Once for each period reported.	The quantity, total duration, and average duration for Outbound calls during the reporting period are presented. A call is considered outbound if the CallType field of the Call table record is set to 3 (Outbound). Duration is calculated here in the same manner as for Inbound calls. Please see that description for details.
Total All - Qty	Once for each period reported.	The quantity, total duration, and average duration of all calls during this period. This simply represents both the Inbound and Outbound columns. Inbound and Outbound quantity and total duration are added together and then averaged.

Table B-4 User Activity Summary Report field descriptions (Continued)

Field	Presence/ Frequency	Description
Outbound Non-Local Trunk - Qty	Once for each period reported.	The quantity, total duration, and average duration for Outbound non-local calls during the reporting period are presented. A call is considered outbound if the CallType field of the Call table record is set to 3 (Outbound). A call is considered Non-Local if the LongDistance field in the Call table is set to true. This flag indicates whether or not the call was long distance from the perspective of the trunk that was used for the call. The calls reported here, are a subset of the calls reported under Outbound all. Duration is calculated in the same manner as for Inbound calls. Please see that description for details.
Outbound WAN Trunk- Qty	Once for each period reported.	The quantity, total duration, and average duration for Outbound non-local calls during the reporting period are presented. A call is considered outbound if the CallType field of the Call table record is set to 3 (Outbound). A call is considered a WAN call if a media stream was established between 2 sites. This is determined by looking in the MediaStream table for any records of media stream for this call (the MediaStream CallID will equal the CallID in the Call table record for the call. The calls reported here, are a subset of the calls reported under Outbound all. Duration is calculated here in the same manner as for Inbound calls. Please see that description for details.
Grand Total		The totals for all the users in the system.

B.5.2 User Activity Detail Report

The User Activity Detail Report (Figure B-6) shows a list of every call for each user by user. This includes the time a call was received or made, the number dialed, and the trunk used. This report can be run as an interval report in which user activity is sub-totaled by the selected interval.

NOTE Conference calls that use a ShoreTel conferencing device have two entries in the User Activity Detail Report. The first entry shows the amount of time (duration) used to enter a pass code or user prompt. The second entry shows the duration of the entire conference call.

User Activity Detail Report

Start Date: End Date: Show Internal Calls: True

Start Time: 12:00:00AM End Time: 11:59:59PM

Interval: 30 min.

AustinWorkgroup (x162) User Activity

Date	Time	In/Out	WG	WAN	Time Stamp	Action	Dialed #	Calling #	Trunk	Duration
	11:00 - 11:29									
06/28/2010	11:10:24AM	In-Int	Yes	N-NS	11:10:24AM	Called	162	134		0:04:20
06/28/2010	11:10:41AM	In-Int	Yes	N-NS	11:10:41AM	Called	162	134		0:04:36
06/28/2010	11:11:49AM	In-Int	Yes	N-NS	11:11:49AM	Called	162	121		0:00:18
06/28/2010	11:18:48AM	In-Int	Yes	N-NS	11:18:48AM	Called	162	121		0:00:26
06/28/2010	11:19:27AM	In-Int	Yes	N-NS	11:19:27AM	Called	162	121		0:00:02
06/28/2010	11:19:43AM	In-Int	Yes	N-NS	11:19:43AM	Called	162	121		0:00:23
06/28/2010	11:20:53AM	In-Int	Yes	N-NS	11:20:53AM	Called	162	121		0:00:20
06/28/2010	11:21:57AM	In-Int	Yes	N-NS	11:21:57AM	Called	162	121		0:00:15
06/28/2010	11:22:52AM	In-Int	Yes	N-NS	11:22:52AM	Called	162	121		0:00:15
	Sub Total							9 Call(s)	0:10:55 Total	0:01:12 Average

Figure B-6 User Activity Detail Report

B.5.2.1 Calls Included

The User Activity Detail Report always displays External Calls and can be configured to display Internal Calls. External calls are those calls where the record in the Call table has a CallType of 2 (Inbound) or 3 (Outbound). All specified calls that have at least one leg with a TalkTime greater than zero are included in the report.

B.5.2.2 Name (Extension Field)

For outbound calls, the Name (Extension) field of the Call record always reports the party that initiated the call.

Inbound calls are reported according to the last party involved in the call (excluding voice mail and the auto-attendant). For example, if a call to extension 320 is not answered and the user's Call Handling Mode (CHM) forwards the call to his or her assistant at extension 452 who answers the call, the Extension field in the Call record contains 452.

When an inbound call is forwarded to voice mail, the Name (Extension) field records the party involved in the call before it was forwarded to voice mail. For example, if a user with extension 320 doesn't answer a call and his or her Call Handling Mode (CHM) forwards the call to voice mail, the extension field is set to 320.

The User Activity Detail Report is described in Table B-5.

Table B-5 User Activity Detail Report Field Descriptions

Field	Presence/ Frequency	Description
Name (Extension)	Once for each extension reported.	The Last Name, First Name, and Extension being reported upon. These come from the PartyIDLastName, PartyIDName, and PartyID for the Connect record that matches the extension in the Call table. Non-users such as Workgroups, Voice Mail, Voice Mail Login, and Auto-attendant are included in the report. The names for these extensions are reported for calls that only interact with these extensions (not a user extension). Like many other non-user extensions, the ShoreTel Audio Conference extension is also not displayed
Date/Time	Once for each call reported.	The date and time when the call being reported started. These fields come from the StartTime field in the Call table record for the call being reported. When interval reports are generated, the actual time the call started is reported even if the call continues into another interval.
In/Out	Once for each call reported.	If the CallType field of the Call record for the call is 2 (Inbound), "In-Int" is shown for internal calls and "In-Ext" for external calls. If the CallType is 3 (Outbound), "Out-Int" is shown for internal calls and "Out-Ext" is shown for external calls. ShoreTel Audio Conference Service calls are displayed as if the service called the user, this field will be 'In-Int' for ShoreTel Audio Conference Service
WG (Workgroup)	Once for each call reported.	The Workgroup field of the Call record for the call is examined. "Yes" or "No" is displayed depending upon whether or not the field indicates a workgroup call.
WAN	Once for each call reported.	A call is considered a WAN call if a media stream was established between 2 sites. This is determined by looking in the MediaStream table for any media stream records for this call. The MediaStream CallID will equal the CallID in the Call table record for the call. Based on the CDR data, the following are displayed: • "Y" or "N" (Y = Yes, N = No) • "S" or "NS" (S = Secure, NS = Not Secure) • "N-S" or "Y-S" (N-S = No, Secure; Y-S = Yes, Secure) • "N-NS" or "Y-NS" (N-NS = No, Not Secure; Y-NS = Yes, Not Secure) • "N-VPN-S" or "Y-VPN-S" (N-VPN-S = No, Virtual Private Network, Secure) (Y-VPN-S = Yes, Virtual Private Network, Secure) • "N-VPN-NS" or "Y-VPN-NS" (N-VPN-NS = No, Virtual Private Network, Not Secure) (Y-VPN-NS = Yes, Virtual Private Network, Not Secure)

Table B-5 User Activity Detail Report Field Descriptions (Continued)

Field	Presence/ Frequency	Description
User Activity Time Stamp Action	Once for each call reported.	This gives the action taken on the call and the time when the action took place. For example, in case of a transfer, the exact time of transfer is reported.
Dialed #	Once for each call reported.	For outbound calls, this is the number the user dialed and is reported in full, canonical format (including country code). For inbound calls, this is the destination of the call. If the call was a DID or DNIS call, this is the DID or DNIS information for the number dialed. For other types of calls, this is the extension where the call first terminates. The dialed number is retrieved from the Dialed Number field of the Call table record for the call.
Calling #	Once for each call reported.	For inbound calls, this is the calling number—ANI or Caller ID—received by the ShoreTel system and is reported as delivered by the PSTN (may or may not include the 1 in front of the area code). The dialed number is retrieved from the CallerID field of the Call table record. For outbound calls, this is the extension of the user that placed the call. In the case of Outbound calls, this data is retrieved from the PartyID field of the Connect record for the party that initiated the call.
Trunk	Once for each call reported.	This is the first trunk that was used for the call. This data is retrieved from the PortName field of the Connect record for the trunk's involvement in the call.
Duration		The duration of the call at the indicated extension. Duration is from the time media is established till it ends. Ring time is not considered in IN/OUT calls. Duration should match Talktime+Hold time (if talk time is non-zero value) This applies to all the reports. This data is retrieved from the Duration field of the Connect table's record for this connection (where the Connect.CallTableID matches the Call.ID and Connect. PartyID matches Call.Extension).
Total		The total number of calls, total duration, and average duration for the user.
Grand Total		The total number of calls, total duration, and average duration for all users.

B.5.3 Trunk Activity Summary Report

The Trunk Activity Summary Report (Figure B-7) shows a summary of all calls for each trunk by trunk group. This includes the type of calls, durations, and average durations made on each trunk.

Trunk Activity Summary Report

Starting Date: 12:00:00AM Ending Date: 11:59:59PM

	Inbound			Outbound			Total		
Bandwidth	Qty	Duration	Average Duration	Qty	Duration	Average Duration	Qty	Duration	Average Duration
Bandwidth 41	4	0:03:14	0:00:48	0	0:00:00	0:00:00	4	0:03:14	0:00:48
Bandwidth 45	4	0:05:29	0:01:22	0	0:00:00	0:00:00	4	0:05:29	0:01:22
Bandwidth 46	36	0:12:46	0:00:21	0	0:00:00	0:00:00	36	0:12:46	0:00:21
Total	**44**	**0:21:29**	**0:00:29**	**0**	**0:00:00**	**0:00:00**	**44**	**0:21:29**	**0:00:29**

	Inbound			Outbound			Total		
Fremont ALS	Qty	Duration	Average Duration	Qty	Duration	Average Duration	Qty	Duration	Average Duration
5109799439	102	0:23:28	0:00:13	0	0:00:00	0:00:00	102	0:23:28	0:00:13
5109799831	2	0:00:10	0:00:05	0	0:00:00	0:00:00	2	0:00:10	0:00:05
Total	**104**	**0:23:38**	**0:00:13**	**0**	**0:00:00**	**0:00:00**	**104**	**0:23:38**	**0:00:13**

Figure B-7 Trunk Activity Summary Report

B.5.3.1 Calls Included

Any trunk activity is reported in the Trunk Activity Summary Report. These calls always have Call records with CallType of 2 (Inbound), 3 (Outbound), or 4 (Tandem). However, with ShoreTel, Release 2.0 and greater, the report is based on the TrunkDirection field in the Connect records representing trunk usage, not the CallType record. There is a 30-day change-over period for upgrades, during which the report reflects both methods of collecting this data.

The Trunk Activity Summary Report is described in Table B-6.

Table B-6 Trunk Activity Summary Report Field Descriptions

Field	Presence/ Frequency	Description
Trunk Group Name	Once for each TrunkGroup being reported upon.	This is the name of the TrunkGroup being reported upon. It's retrieved from the GroupName field of the Connect record representing a trunk's involvement in a call.
Trunk Name	Once for each trunk being reported upon.	The name of the specific trunk used for a call. This is retrieved from the PortName field of the Connect record representing a trunk's involvement in a call.
Inbound Qty, Duration, and Average Duration	Once for each trunk being reported upon.	The quantity, total duration, and average duration for all inbound trunk activity for this trunk during the reporting period are presented. Trunk activity is considered inbound if the TrunkDirection field in the Connect record is set to 2 (Inbound). The quantity is simply a count of the Connect table records during the reporting period for this trunk that indicate inbound trunk usage. Duration is the sum of all the Duration fields for the Connect table records during the reporting period for this trunk that indicate inbound trunk usage. The average duration is found by dividing this total by the quantity reported here.

Table B-6 Trunk Activity Summary Report Field Descriptions (Continued)

Field	Presence/ Frequency	Description
Outbound Qty, Duration, and Average Duration	Once for each trunk being reported upon.	The quantity, total duration, and average duration for all outbound trunk activity for this trunk during the reporting period are presented. Trunk activity is considered outbound if the TrunkDirection field in the Connect record is set to 3 (Outbound). The quantity is simply a count of the Connect table records during the reporting period for this trunk that indicate outbound trunk usage. Duration is the sum of all the Duration fields for the Connect table records during the reporting period for this trunk that indicate outbound trunk usage. Average duration is found by dividing this total by the quantity reported here.
Total Qty	Once for each trunk being reported upon.	The total calls for the trunk.
Duration	Once for each trunk being reported upon.	The total duration of calls, in hours, minutes, and seconds.
Average Duration	Once for each trunk being reported upon.	The average duration of calls, in hours, minutes, and seconds.
Total		The totals for all trunks in the trunk group.
Grand Total		The totals for all trunks in the system.

B.5.4 Trunk Activity Detail Report

The Trunk Activity Detail Report (Figure B-8) shows a list of every call for each trunk by trunk group. This includes the date and time, the number dialed, and the user's name.

Trunk Activity Detail Report

Starting Date: 12:00:00AM Ending Date: 11:59:59PM

Austin ALS	Date	Time	In/Out	Dialed #	Calling #	User	Duration
1	06/28/2010	2:33:47AM	Out	9+919910487036			0:00:00
	Sub-Total				1 Call(s)	0:00:00 Average	0:00:00 Total
Total					1 Call(s)	0:00:00 Average	0:00:00 Total

E1 to France	Date	Time	In/Out	Dialed #	Calling #	User	Duration
E1 to France 2	07/19/2010	7:18:28PM	Out	324	131	User1	0:00:00
E1 to France 2	07/19/2010	7:18:43PM	Out	324	131	User1	0:00:05
E1 to France 2	07/19/2010	7:24:37PM	Out	323	131	User1	0:00:00
E1 to France 2	07/19/2010	8:19:40PM	Out	323	131	User1	0:00:00
E1 to France 2	07/19/2010	8:19:55PM	Out	323	131	User1	0:00:00
E1 to France 2	07/19/2010	8:26:35PM	Out	14089623240	+15109283596		0:00:00
	Sub-Total				6 Call(s)	0:00:00 Average	0:00:05 Total
E1 to France 5	07/19/2010	7:33:40PM	In	14089621175	323	User1	0:00:02
E1 to France 5	07/19/2010	7:35:12PM	In	14089621175	Unknown	User1	0:01:11
E1 to France 5	07/19/2010	8:26:35PM	In	14089623240	Unknown	User1	0:00:06
	Sub-Total				3 Call(s)	0:00:26 Average	0:01:19 Total
Total					9 Call(s)	0:00:09 Average	0:01:24 Total

Figure B-8 Trunk Activity Detail Report

B.5.4.1 Calls Included

Any trunk activity is reported in the Trunk Activity Detail Report. These calls always have Call records with CallType of 2 (Inbound), 3 (Outbound), or 4 (Tandem). However, with ShoreTel Release 2.0 and greater, the report is based on the TrunkDirection field in the Connect records representing trunk usage, not the CallType record. There is a 30-day crossover period for upgrades during which the report will reflect both methods of collecting this data.

The Trunk Activity Detail Report is described in Table B-7.

Table B-7 Trunk Activity Detail Report Field Descriptions

Field	Presence/Frequency	Description
Trunk Group Name	Once for each TrunkGroup being reported upon. If the trunk activity for a TrunkGroup requires more than one page to report, then the name is repeated at the top of each additional page.	This is the name of the TrunkGroup being reported upon. This data is retrieved from the GroupName field of the Connect record representing a trunk's involvement in a call.
Trunk Name	Once for each TrunkGroup being reported upon. If the trunk activity for a TrunkGroup requires more than one page to report, then the name is repeated at the top of each additional page.	The name of the specific trunk used. This data is retrieved from the PortName field of the Connect record representing a trunk's involvement in a call.
Date	Once for each trunk activity in the report.	The date is extracted from the ConnectTime field in the Connect record representing the trunk's involvement in the call. This is the date the trunk was added to the call.
Time	Once for each trunk activity in the report.	The time is extracted from the ConnectTime field in the Connect record representing the trunk's involvement in the call. This is the time the trunk was added to the call.
In/Out	Once for each trunk activity in the report.	Trunk activity is considered “In” if the TrunkDirection field in the Connect record is set to 2 (Inbound), otherwise it is considered “Out.” When an external user calls the external number of the UCB, two records are displayed in the report, and each record is considered “In.”

Table B-7 Trunk Activity Detail Report Field Descriptions (Continued)

Field	Presence/Frequency	Description
Dialed #	Once for each trunk activity in the report.	For outbound calls, this is the number the user dialed and is reported in full, canonical format (including country code, etc.). For an inbound call this is the destination of the call. If the call was a DID or DNIS call, this is the DID or DNIS information for the number dialed. For other types of calls, this is the extension where the call first terminates. For inbound calls (CallType = 2 in Call record) this data is retrieved from the DialedNumber field in the Call record. For other calls, it is retrieved from the PartyId field of the Connect record for the trunk activity. Inbound and outbound are relative to the call, not trunk usage.
Calling #	Once for each trunk activity in the report.	For inbound calls, this is the calling number—ANI or Caller ID—received by the ShoreTel system and is reported as delivered by the PSTN (may or may not include the 1 in front of the area code). For outbound calls, this is the extension of the user that placed the call. For outbound calls (CallType = 3 in Call record) this data is retrieved from the Extension field in the Call record. For other types of calls, it is retrieved from the CallerID field in the Call table. Inbound and outbound are relative to the call, not trunk usage.
User	Once for each trunk activity in the report.	The name associated with the extension that was the initial target of the call. For outbound calls (CallType = 3 in Call record), the user is the extension that first initiated the call. For inbound calls (CallType = 2 in Call record), the user is the extension that was the initial target of the call. In the case of Tandem calls (CallType = 4 in the Call record), nothing is shown. This data is retrieved from the PartyIDName and PartyIDLastName fields of the Connect record for the party that initiated the call (ConnectReason = 19, "Originate" for an outbound call) or was the target of the call (ConnectReason = 17, "Called" for inbound call). Inbound and outbound are relative to the call, not trunk usage.
Duration	Once for each trunk activity in the report.	The duration of the trunk activity. This data is retrieved from the Duration field of the Connect record for the trunk's involvement in the call. For an inbound call, the duration of the call begins when the trunk is seized and includes the ring time, talk time, and hold time. The duration ends when the user hangs up or when the external party hangs up and disconnect supervision is received by the ShoreTel system. For an outbound call, the duration of the call begins when the trunk is seized. The duration ends when the user hangs up, or when the external party hangs up and disconnect supervision is received by the ShoreTel system.

Table B-7 Trunk Activity Detail Report Field Descriptions (Continued)

Field	Presence/Frequency	Description
Subtotal		The total of calls for the trunk.
Total		The total of calls for the trunk group.
Grand Total		The total of calls for all trunks in the system.

B.5.5 Workgroup Agent Summary Report

The Workgroup Agent Summary Report (Figure B-9) shows a summary of inbound workgroup calls and agent activity by the workgroup.

Workgroup Agent Summary Report

Start Date: | End Date:

Start Time: 12:00:00AM | End Time:11:59:59PM | Duration Format: hh:mm:ss

	Inbound Workgroup Calls			Inbound User Calls			Outbound Calls			Other Calls			Total Calls			Agent Activity		
	Qty	Duration	Average Duration	Qty	Duration	Average Duration	Qty	Duration	Average Duration	Qty	Duration	Average Duration	Qty	Duration	Average Duration	Total Wrapup	Average Wrapup	Total Login
NAS Workgroup (x1000)																		
Alexanian, Ed (x3376)																		
11/14/2008 (Fri)	0	00:00:00	00:00:00	0	00:00:00	00:00:00	1	00:00:55	00:00:55	0	00:00:00	00:00:00	1	00:00:55	00:00:55	00:00:00	00:00:00	17:20:35
3/17/2009 (Tue)	0	00:00:00	00:00:00	0	00:00:00	00:00:00	1	00:00:32	00:00:32	0	00:00:00	00:00:00	1	00:00:32	00:00:32	00:00:00	00:00:00	24:00:00
7/7/2009 (Tue)	0	00:00:00	00:00:00	0	00:00:00	00:00:00	0	00:00:00	00:00:00	1	00:05:22	00:05:22	1	00:05:22	00:05:22	00:00:00	00:00:00	24:00:00
10/30/2009 (Fri)	0	00:00:00	00:00:00	0	00:00:00	00:00:00	1	00:01:21	00:01:21	0	00:00:00	00:00:00	1	00:01:21	00:01:21	00:00:00	00:00:00	24:00:00
User Sub Total	0	00:00:00	00:00:00	0	00:00:00	00:00:00	3	00:02:48	00:00:56	1	00:05:22	00:05:22	4	00:08:10	00:02:02	00:00:00	00:00:00	89:20:35
Workgroup Subtotal	**0**	**00:00:00**	**00:00:00**	**0**	**00:00:00**	**00:00:00**	**3**	**00:02:48**	**00:00:56**	**1**	**00:05:22**	**00:05:22**	**4**	**00:08:10**	**00:02:02**	**00:00:00**	**00:00:00**	**89:20:35**

Figure B-9 Workgroup Agent Summary Report

B.5.5.1 Calls Included

This report includes calls routed to workgroup agents by the workgroup server, and non-workgroup calls (both inbound and outbound). The report assigns non-workgroup calls to an agent's membership within a workgroup by examining the workgroup the agent was logged into during or before the call. No calls are reported when an agent is logged out. You can find Agent logins by examining the AgentActivity table for records with State = 5 (LogInOut).

Workgroup agents can be a member of more than one workgroup. When they log in, their login time is reported for all workgroups of which they are a member.

Non-workgroup calls are reported against the workgroup with the lowest dial number that the agent is a member of when the call is made. For example, if the agent is a member of workgroups with dial numbers of 1100, 1200, and 1250, non-workgroup calls are reported against 1100.

The StartTimeStamp field in these Agent Activity records represents the time that an agent logged into a specific workgroup (identified by the WorkgroupDN and WorkgroupName fields). The EndTimeStamp field records the time the agent logged out (this can be null when the agent is still logged into the workgroup).

This report is call centric. While it does report agent activity, which consists of agent wrap-up and login time, the report will only show this information for periods during which there was a call for the agent (workgroup or non-workgroup).

The Workgroup Agent Summary Report is described in Table B-8.

Table B-8 Workgroup Agent Summary Report field descriptions

Field	Presence/Frequency	Description
Workgroup Name	Once for each workgroup being reported upon.	The name of the workgroup agent.
Agent Name	Once for each agent reported upon (within an interval if intervals are selected when running the report).	The name of the agent and his or her extension. These come from the AgentActivity table's AgentLastName, Agent-FirstName, and AgentDN fields.
Inbound WG Calls Qty, Duration, and Average Duration	Once for each agent reported upon (within an interval if intervals are selected when running the report).	The quantity, total duration, and average duration for all inbound workgroup calls during the reporting period are presented. Call is considered to be inbound if the CalType is set to 2 (Inbound). The quantity is simply a count of the Connect table records during the reporting period for this workgroup that indicate the call as inbound. Duration is the sum of all the Duration fields for the Connect table records during the reporting period for this workgroup. Average duration is found by dividing this total by the quantity reported here.
Inbound user Calls Qty, Duration, and Average Duration	Once for each agent reported upon (within an interval if intervals are selected when running the report).	The quantity, total duration, and average duration for all inbound user calls during the reporting period are presented. Call is considered to be inbound user if the CalType is set to 2 (Inbound) and workgroup is not set in the call table. The quantity is simply a count of the Connect table records during the reporting period for this user that indicate the call as inbound. Duration is the sum of all the Duration fields for the Connect table records during the reporting period for this user. Average duration is found by dividing this total by the quantity reported here.
Outbound Calls Qty, Duration, and Average Duration	Once for each agent reported upon (within an interval if intervals are selected when running the report).	The quantity, total duration, and average duration for all outbound user calls during the reporting period are presented. Call is considered to be outbound user if the CalType is set to 3 (Outbound) and workgroup is not set in the call table. The quantity is simply a count of the Connect table records during the reporting period for this user that indicate the call as outbound. Duration is the sum of all the Duration fields for the Connect table records during the reporting period for this user. Average duration is found by dividing this total by the quantity reported here.

Table B-8 Workgroup Agent Summary Report field descriptions (Continued)

Field	Presence/Frequency	Description
Other Calls Qty, Duration, and Average Duration	Once for each agent reported upon (within an interval if intervals are selected when running the report).	The quantity, total duration, and average duration for all other calls during the reporting period are presented. Call is considered to be other if the call is neither inbound nor outbound. The quantity is simply a count of the Connect table records during the reporting period. Duration is the sum of all the Duration fields for the records during the reporting period for this user. Average duration is found by dividing this total by the quantity reported here.
Total Calls Qty, Duration, and Average Duration	Once for each agent reported upon (within an interval if intervals are selected when running the report).	The quantity, total duration, and average duration for total calls during the reporting period are presented. It is sum of the all inbound, outbound and other calls. The quantity is simply a count of the Connect table records during the reporting period. Duration is the sum of all the Duration fields for the records during the reporting period for this user. Average duration is found by dividing this total by the quantity reported here.
Agent Activity Total Wrapup, Average Wrapup, and Total Login	Once for each agent reported upon (within an interval if intervals are selected when running the report).	The Total wrapup is simply the count of the wrapup records in the agent activity table during the reporting period for the agent. Average wrapup is found by dividing this total by the quantity reported here. The total login time is the time for which agent is logged in as workgroup agent.

B.5.6 Workgroup Agent Detail Report

The Workgroup Agent Detail Report (Figure B-10) shows a list of every call for each agent by workgroup.

Workgroup Agent Detail Report

Start: 12:00:00AM End: 11:59:59PM

	Date/Time	Dialed #	Calling #	Call Type	Trunk	Call Duration	Wrapup Duration	Queue Duration	Total Duration
June 29, 2010									
AustinWorkgroup (x162)									
15:30 - 15:59									
User12 (x122)									
	6/29/2010 3:50:58PM	14089621175	+14082037940	InWG	JY Shared PRI - 1	00:00:20	00:00:30	00:00:04	00:00:54
Sub Total				**1 Call(s)**	**00:00:20 Average**	**00:00:20**	**00:00:30**	**00:00:04**	**00:00:54**
Sub Total				**1 Call(s)**	**00:00:20 Average**	**00:00:20**	**00:00:30**	**00:00:04**	**00:00:54**
16:00 - 16:29									
User12 (x122)									
	6/29/2010 4:06:10PM	14089621175	+14082037940	InWG	JY Shared PRI - 1	00:00:15	00:00:30	00:00:03	00:00:48
	6/29/2010 4:15:32PM	14089621175	+14082037940	InWG	JY Shared PRI - 1	00:00:30	00:00:30	00:00:04	00:01:04
Sub Total				**2 Call(s)**	**00:00:22 Average**	**00:00:45**	**00:01:00**	**00:00:07**	**00:01:52**
Sub Total				**2 Call(s)**	**00:00:22 Average**	**00:00:45**	**00:01:00**	**00:00:07**	**00:01:52**
Sub Total				**3 Call(s)**	**00:00:21 Average**	**00:01:05**	**00:01:30**	**00:00:11**	**00:02:46**

Figure B-10 Workgroup Agent Detail Report

An agent appears in the Workgroup Agent Detail Report if the agent had any workgroup call activity during the reporting period.

B.5.6.1 Calls Included

This report includes calls routed to workgroup agents by the workgroup server, and non-workgroup calls (both inbound and outbound). The report assigns non-workgroup calls to an agent's membership within a workgroup by examining the workgroup the agent was logged into during or before the call. Non-workgroup calls made while an agent is logged out are not reported.

Workgroup agents can be a member of more than one workgroup. When they log in, their login time is reported for all workgroups of which they are a member.

Non-workgroup calls are reported against the workgroup with the lowest dial number that the agent is a member of when the call is made. For example, if the agent is a member of workgroups with dial numbers of 1100, 1200, and 1250, non-workgroup calls are reported against 1100.

You can find agent logins by examining the AgentActivity table for records with State = 5 (LogInOut). The StartTimeStamp field in these AgentActivity records represents the time that an agent logged into a specific workgroup (identified by the WorkgroupDN and WorkgroupName fields). The EndTimeStamp field records the time the agents logged out (this can be null when the agent is still logged into the workgroup).

The Workgroup Agent Detail Report is described in Table B-9.

Table B-9 Workgroup Agent Detail Report field descriptions

Field	Presence/ Frequency	Description
Workgroup Name	Once for each workgroup being reported upon.	The name of the workgroup agent.
Agent Name	Once for each agent reported upon (within an interval if intervals are selected when running the report).	The name of the agent and his or her extension. These come from the AgentActivity table's AgentLastName, AgentFirstName, and AgentDN fields.
Date/Time	Once for each call reported.	The date and time for the call being reported. These fields come from the StartTime field in the Call table record for the reported call. When interval reports are generated, the call is reported for the time when it starts even if it continues into another interval.
Dialed #	Once for each call reported.	For inbound calls, this is the destination of the call. If the call was a DID or DNIS call, this is the DID or DNIS information for the number dialed. For other types of calls, this is the extension where the call first terminates. The number dialed to initiate the call. This comes from the Call record's DialedNumber field.
Calling #	Once for each call reported.	The caller that initiated the call. For inbound calls, this is the calling number—ANI or Caller ID—received by the ShoreTel system and is reported as delivered by the PSTN (may or may not include the "1" in front of the area code). This comes from the Call table's CallerID field.

Table B-9 Workgroup Agent Detail Report field descriptions (Continued)

Field	Presence/ Frequency	Description
Call Type	Once for each call reported.	Indicates whether this is an incoming workgroup call ("InWG"), an inbound non-workgroup call ("In"), or an outbound call ("Out"). A call is categorized as an inbound workgroup call if the user joined the call as a workgroup agent. This is determined by examining the Connect record for the user's time on the call. The PartyType in the Connect record must be 12 (Workgroup Agent). A call is categorized as an inbound non-workgroup call if the Call record's CallType = 1 or 2 (internal or inbound), and the user's Connect record has PartyType = 1 (station), and the user was not the originator of the call (ConnectReason in the Connect table not equal to 19-- originate). A call is categorized as outbound if the user originated the call (as shown by the Connect record having ConnectReason = 19). These calls have CallType = 1 or 3 (internal or outbound) in the call record. Note that for all calls, those calls with CallType = 1 (internal) are included only if the option to include internal calls is chosen. Calls that involve multiple legs are also reported as: Transfer—Call was transferred. Conference—Call was conferenced Monitor—Call was monitored Barge-In—Call was barged
Other Calls	Once for each call reported.	Indicates calls that do not belong to Call Type (i.e. those that are no In, InWG, or Out). This includes: Transfer—Call was transferred. Conference—Call was conferenced Monitor—Call was monitored Barge-In—Call was barged
Trunk	Once for each call reported.	This is the first trunk that was used for the call. This data is retrieved from the PortName field of the Connect record for the trunk's involvement in the call. In the case of calls not involving a trunk, it will be blank (this can occur with internal calls).
Call Duration	Once for each call reported.	The duration of the call. Duration reports the time of the user's involvement in the call. It's reported by summing the TalkTime, RingTime, and HoldTime fields in the Connect record representing involvement in the call. Since a call is reported during the period in which it starts (as identified by the StartTime in the Call table record for the call) but may end during another interval, the duration can be longer than the 30- or 60-minute interval period. The total duration is reported during the interval in which the call begins.

Table B-9 Workgroup Agent Detail Report field descriptions (Continued)

Field	Presence/ Frequency	Description
Wrap-Up Duration	Once for each call reported.	Only applicable to inbound workgroup calls. This is the amount of time, if any, the agent spent in wrap-up after completing the call. The Wrap-up Duration is the difference between the StartTimeStamp and EndTimeStamp in the wrap-up record in the AgentActivity table for that agent.
Queue Duration	Once for each call reported.	Only applicable to inbound workgroup calls, the amount of time that the call was in the workgroup queue before it was assigned to the agent. This data is retrieved from the Duration field of the QueueCall table record for the call.
Total Duration	Once for each call reported.	The Total Duration includes the Queue Duration, Call Duration, and Wrap-up Duration. It is generally less than the total time the call spends within the ShoreTel system. The time between when the trunk was seized and the call was accepted by the workgroup, or any time the call spends with a menu or other extension, is not reflected.
Sub-total		The total calls for the workgroup agent.
Total		The total calls for all agents in the workgroup.
Grand Total		The total calls for all agents in the system.

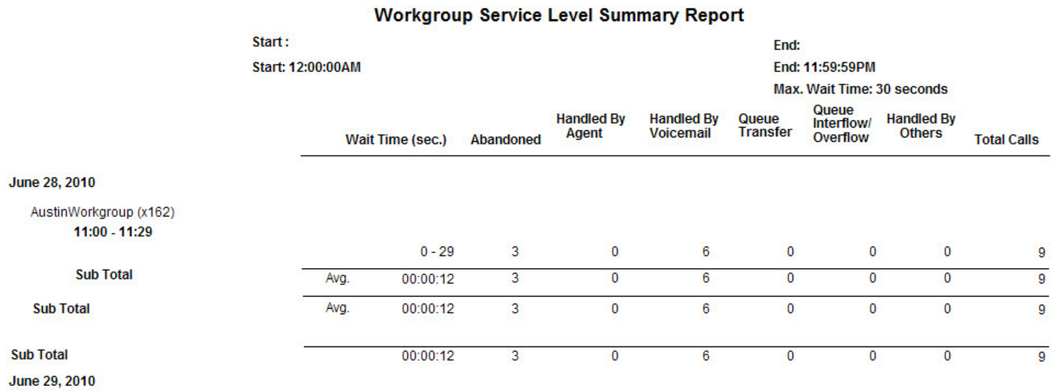
B.5.7 Workgroup Service Level Summary Report

The Workgroup Service Level Summary Report provides information related to the workgroup server call processing. Every time the workgroup server processes a call, a record about its disposition is added to the QueueCall table. Generally, this occurs once when the call gets processed by the server. However, in the case of call forwarding, the same call can pass through the workgroup server more than once.

For example, a call made to the workgroup server is transferred to an extension. If that extension's call handling mode forwards the call to the same or a different workgroup, the call passes through the workgroup server more than once. The rule in these cases is simple—for each time the workgroup server disposes of a call, a record is added to the QueueCall table.

The report always includes external calls to a workgroup. Internal workgroup calls are included in the report only if the option to include them is selected (the default is *not included*). The CallType field in the Call table is examined to determine if the call is internal or external. If the CallType is 1 (extension to extension), it is an internal call; otherwise it is an external call. The QueueCall record for a call processed by the workgroup server has a ConnectTableID that identifies the Connect table entry for the workgroup server being added to the call. The Connect table entry has a CallTableID field that is examined to determine the Call table record for the call, whether the call is internal or external.

If the workgroup service is not operational, the call is not processed by the workgroup server (it simply goes to the backup extension). Calls directed to the workgroup but not processed by the service because it is down are not included in this report. When this occurs, there is no record of the call in the QueueCall table since records are only added to the table when the workgroup server processes the call. Figure B-11 is an example of a Workgroup Service Level Summary Report.



Workgroup Service Level Summary Report

Start :
Start: 12:00:00AM
End:
End: 11:59:59PM
Max. Wait Time: 30 seconds

		Wait Time (sec.)	Abandoned	Handled By Agent	Handled By Voicemail	Queue Transfer	Queue Interflow/ Overflow	Handled By Others	Total Calls
June 28, 2010									
AustinWorkgroup (x162) 11:00 - 11:29									
		0 - 29	3	0	6	0	0	0	9
Sub Total	Avg.	00:00:12	3	0	6	0	0	0	9
Sub Total	Avg.	00:00:12	3	0	6	0	0	0	9
Sub Total		00:00:12	3	0	6	0	0	0	9
June 29, 2010									

Figure B-11 Workgroup Service Level Summary Report

The Workgroup Service Level Summary Report is described in Table B-10.

Table B-10 Workgroup Service Level Summary Report field descriptions

Field	Presence/ Frequency	Description
Wait Time (sec.)	Shown once for each 30-second period in workgroup/ internal where there are calls to be reported.	Range of wait-for-service times, The wait time is divided into 30-second intervals. Information for the calls is reported for the interval in which it falls, according to when the call moved off the workgroup. The actual wait or service time for each workgroup call is found in the Duration field of the QueueCall table. This duration is the time from when the call is offered to the workgroup server until it leaves the call queue.
Abandoned	Once for each period reported.	Number of callers who abandoned the call (hung up) during the period. Those QueueCall records with the ExitReason set to 7 (Abandoned) are counted as abandoned.
Handled by Agent	Once for each period reported.	Number of calls handled by agents during the period. Those QueueCall records with the ExitReason set to 1 (TransferToAgent) are counted as handled by an agent. A call that is picked up or unparked by an agent that is a member of the same workgroup is also counted as Handled by Agent.
Handled by Voice Mail	Once for each period reported.	Number of calls that went to the workgroup's voice mail (either as a result of call handling, or when the caller chose the transfer to voice mail option). Those QueueCall reports with the TargetType set to 3 (mailbox) and ExitReason set to 2, 3, 4, or 5 (ForwardAlways, ForwardBusy, ForwardNoAnswer, or ForwardNoLoginAgent), or 9 (TransferVM) with the TargetDN field equal to the workgroup DN itself are counted as Handled by Voice Mail.
Transfer	Once for each period reported.	Number of calls transferred by WorkGroup agents.

Table B-10 Workgroup Service Level Summary Report field descriptions

Field	Presence/ Frequency	Description
Interflow / Overflow	Once for each period reported.	Number of automatic call transfers, based on caller wait time to a dialable number (interflow) or to another WorkGroup queue (overflow).
Handled by Others	Once for each period reported.	Number of calls handled by others (not workgroup agents or voice mail). Any call for the workgroup that isn't reported as Abandoned, Handled by Agent, or Handled by Voice Mail is counted as Handled by Others. Calls that are picked up by non-agents, or agents that do not belong to the group are counted as Handled by Others.
Total Calls	Once for each period reported.	Sum of Abandoned, Handled by Agent, Handled by Voice Mail, Picked Up from the Queue, Unparked from the Queue, and Handled by Others for the period.

B.5.8 Workgroup Queue Summary Report

The Workgroup Queue Summary Report (Figure B-12) is a summary of queue activity and how the calls interact with the queue. The report can run with fixed interval sub-totals.

Workgroup Queue Summary Report

Start:
Start:12:00:00AM

End:
End:11:59:59PM
Interval: 30 min.

	Abandoned	Handled by Agent	Handled by WG Voicemail	Queue Transfer	Queue OverFlow/ Interflow	Handled by Others	Maximum Abandon Time	Average Abandon Time	Maximum Handled Time	Average Handled Time	Total Calls
AustinWorkgroup (x162)											
11:00 - 11:29 06/28/2010 (Mon)	3	0	6	0	0	0	00:00:23	00:00:14	00:00:24	00:00:12	9
	3	0	6	0	0	0	00:00:23	00:00:14	00:00:24	00:00:12	9
15:00 - 15:29 07/09/2010 (Fri)	0	0	5	0	0	0	00:00:00	00:00:00	00:00:53	00:00:15	5
	0	0	5	0	0	0	00:00:00	00:00:00	00:00:53	00:00:15	5
15:30 - 15:59 06/29/2010 (Tue)	0	1	0	0	0	0	00:00:00	00:00:00	00:00:04	00:00:04	1
	0	1	0	0	0	0	00:00:00	00:00:00	00:00:04	00:00:04	1
16:00 - 16:29 06/29/2010 (Tue)	2	2	0	0	0	0	00:00:07	00:00:04	00:00:04	00:00:03	4
	2	2	0	0	0	0	00:00:07	00:00:04	00:00:04	00:00:03	4
17:30 - 17:59 06/29/2010 (Tue)	1	0	0	0	0	0	00:00:02	00:00:02	00:00:00	00:00:00	1
	1	0	0	0	0	0	00:00:02	00:00:02	00:00:00	00:00:00	1
21:00 - 21:29 07/19/2010 (Mon)	3	1	3	0	0	0	00:00:35	00:00:16	00:00:37	00:00:16	7
	3	1	3	0	0	0	00:00:35	00:00:16	00:00:37	00:00:16	7
Sub Total	9	4	14	0	0	0	00:00:35	00:00:11	00:00:53	00:00:12	27

Figure B-12 Workgroup Queue Summary Report

B.5.8.1 Calls Included

The key determinant in this report is which workgroup server processes the call. Each time the workgroup server processes a call, a record about the call's disposition is added to the QueueCall table. In most cases the call is recorded just once, but if forwarded, a call can be recorded twice. Normally the call comes in and is processed by the server where it is routed to an agent. The caller then chooses to go to voice mail or another destination, or hangs up (abandons the call) before it is routed beyond the workgroup. Since the report shows how the call was disposed of by the workgroup server, the call is reported once in the report. However, if the call is forwarded, the same call can pass through the workgroup server more than once.

For example, a call goes to a workgroup server. While on the call, the user transfers it to another extension. The user's extension's call handling mode forwards the call to the same or a different workgroup. In this case, the call passes through the workgroup server more than once and is reported each time the workgroup server processes the call. For each time the workgroup server processes the call, a record is added to the QueueCall table.

External calls to a workgroup are always included in the report. Internal workgroup calls are only included in the report if the option to include them is enabled (by default they are not). The CallType field in the Call table is examined to determine if the call is internal or external. If the CallType is 1 (extension to extension), it is an internal call; otherwise it is an external call. The QueueCall record for a call processed by the workgroup server has a ConnectTableID that identifies the Connect table entry for the workgroup server being added to the call. The Connect table entry has a CallTableID field that is then examined to determine the Call table record for the call. It is this record's CallType that is examined to determine whether the call is internal or external.

If the workgroup service is not operational, the call is not processed by the workgroup server (it simply goes to the backup extension). These calls are not included in the report. When this occurs, there is no record of the call in the QueueCall table, since records are only added to that table when the workgroup server processes the call.

The Workgroup Queue Summary Report is described in Table B-11.

Table B-11 Workgroup Queue Summary Report Field Descriptions

Field	Presence/ Frequency	Description
Workgroup Name		The name of the workgroup.
Calls Abandoned	Once for each period reported.	The number of callers who hung up or otherwise disconnected while waiting in queue. Those QueueCall records with the ExitReason set to 7 (abandoned) are counted as Abandoned.
Calls Handled by Agent	Once for each period reported.	The number of calls that were answered by agents in the workgroup. Those QueueCall records with the ExitReason set to 1 (TransferToAgent) are counted as Handled by Agent.
Calls Handled by Voice Mail	Once for each period reported.	Number of calls that went to the workgroup's voice mail (either as a result of call handling or when the caller chose to transfer to voice mail). Those QueueCall records with the TargetType set to 3 (Mailbox), and ExitReason set to 2, 3, 4, or 5 (ForwardAlways, ForwardBusy, ForwardNoAnswer, or ForwardNoLoginAgent) or 9 (TransferVM) with the TargetDN field equal to the workgroup DN itself are counted as Handled by Voice Mail.

Table B-11 Workgroup Queue Summary Report Field Descriptions (Continued)

Field	Presence/ Frequency	Description
Transfer	Once for each period reported	Number of calls transferred by WorkGroup agents.
Overflow / Interflow	Once for each period reported.	Number of automatic call transfers, based on caller wait time to a dialable number (interflow) or to another WorkGroup queue (overflow).
Calls Handled by Others	Once for each period reported.	Number of calls handled by others (not workgroup's agents or voice mail). Any call for the workgroup that isn't reported as Abandoned, Handled by Agent, Picked Up from the Queue, Unparked from the Queue, or Handled by Voice Mail is counted as Handled by Others.
Maximum Abandoned Time	Once for each period reported.	The maximum time during the period that a caller who abandoned the call stayed on the line. The QueueCall records are examined for calls reported as abandoned during this period. The DurationSeconds field with the largest value for abandoned calls is reported.
Average Abandoned Time	Once for each period reported.	The average time during the period that those callers who abandoned the call stayed on the line. The sum of the DurationSeconds field in all of the QueueCall records for abandoned calls during the period divided by the number of such calls to report the average time abandoned.
Maximum Handled Time	Once for each period reported.	The maximum time during the period that a caller stayed on the line before the call was handled (by agent, voice mail, or others). Note that the maximum time could be zero even though there were handled calls in the case of the call being forwarded immediately to voice mail. The QueueCall records are examined for calls reported as handled during this period. The DurationSeconds field with the largest value for abandoned calls is reported.
Average Handled Time	Once for each period reported.	The average time during the period that a caller was on the line before the call was handled (by an agent, voice mail, or others). Note that the average time could be zero even though there were handled calls in the case of the call being forwarded immediately to voice mail. The sum of the DurationSeconds field in all of the QueueCall records for handled calls during the period divided by the number of such calls to report the average time abandoned.
Total Calls	Once for each period reported.	All calls passed through the workgroup. This includes calls that go straight to agents without waiting in queue. Sum of Abandoned, Handled by Agent, Handled by Voice Mail, and Handled by Others for the period.

B.5.9 WAN Media Stream Summary Report

The WAN Media Stream Summary Report (Figure B-13) shows the summary of call quality and call traffic for calls made over the WAN in multi-site deployments. By understanding the amount of time the WAN is used for calls, you can estimate the amount of toll charges your organization is saving. In addition, by understanding the jitter and packet loss, you can get an approximation of the quality of the WAN link and use this to influence your service provider if required.

The Media Stream Report lists a matrix of all sites and the links to other sites on the system. Media streams are reported rather than calls, since this report focuses on the exact amount of bandwidth used. Calls can be quite complex involving multiple parties, including users, voice mail, and auto-attendant. Each media stream that is reported includes the associated Call ID (Call Identification) that can be correlated to the parties on the call for troubleshooting purposes using the CDR database. Figure B-13 is an example of the Media Stream Summary Report.

Media Stream Summary Report

Starting: 12:00:00AM Ending: 11:59:59PM All Calls

		Quality			Traffic Volume			
Site A	**Site B**	**Avg Jitter (ms)**	**Max Jitter (ms)**	**% Packets Lost**	**Blocked Call**	**Total**	**Duration**	**Avg Duration**
Austin	Austin	2.47	11.00	0.03	0	15	00:06:51	00:00:27
Austin	Germany	2.75	11.00	0.10	0	8	00:04:13	00:00:31
Austin	Guam	6.50	89.00	0.02	0	58	00:46:04	00:00:47
Germany	Germany	0.00	0.00	0.00	0	2	00:00:05	00:00:02
Guam	Germany	4.13	21.00	0.00	0	8	00:05:15	00:00:39
Guam	Guam	0.33	4.00	0.00	0	18	00:34:16	00:01:54
Headquarters	Austin	14.53	333.00	0.01	0	30	00:15:47	00:00:31
Headquarters	Germany	6.38	80.00	0.01	0	13	00:23:59	00:01:50
Headquarters	Guam	4.51	117.00	0.01	0	71	01:37:11	00:01:22
Headquarters	Headquarters	2.10	18.00	0.01	0	63	00:55:04	00:00:52
Headquarters	Paris	0.44	1.00	0.10	0	9	00:01:01	00:00:06
Headquarters	Unknown Site				1	0		

Figure B-13 Media Stream Summary Report

B.5.9.1 Calls Included

This report summarizes media streams (not calls) between the two sites. Media streams can be for extensions or trunks. You can configure the report to display information for all calls or for only intersite calls. IP phone media streams are not included in this report.

The Media Stream Summary Report is described as follows:

Table B-12 WAN Media Stream Summary Report Field Descriptions

Field	Presence/ Frequency	Description
Site A	Once for each pair of sites being reported.	The name of the site. This data is retrieved from the ASiteName field in the MediaStream table.
Site B	Once for each pair of sites being reported.	The name of the site that communicates to Site A. This data is retrieved from the BSiteName field in the MediaStream table.
Quality - Avg Jitter	Once for each pair of sites being reported.	The average jitter is the average of the per-media stream maximum jitter between the sites given in milliseconds. This data is retrieved from the A MaxJitter and B MaxJitter fields in the MediaStream table.

Table B-12 WAN Media Stream Summary Report Field Descriptions (Continued)

Field	Presence/ Frequency	Description
Quality - Max Jitter	Once for each pair of sites being reported.	The maximum jitter is the worst jitter encountered on any media stream between the sites given in milliseconds. This data is retrieved from the A MaxJitter and B MaxJitter fields in the MediaStream table. The jitter buffer should be larger than this value for proper operation. The 'max jitter' value in this report is only recorded up to the maximum jitter buffer value configured in Director.
Quality - % Packet Loss	Once for each pair of sites being reported.	This is the number of packets that were expected to arrive but did not arrive at the destination. Lost packets were mostly likely dropped on their way through the network.
Quality - Blocked Calls	Once for each pair of sites being reported.	The number of media that were not routed across the WAN due to insufficient WAN bandwidth (admission control reached). This could indicate that more WAN bandwidth is required. This is a count of the number of records in the MediaStream table between the two sites with FailureCode = 1 (Admission Control Inhibited Call).
Traffic Volume - Total	Once for each pair of sites being reported.	The number of media streams used between the two sites as recorded in the MediaStream table.
Traffic Volume - Duration	Once for each pair of sites being reported.	The duration of all the media streams used between the two sites. The value is the sum of duration for all records between the two sites in the MediaStream table.
Traffic Volume - Avg Duration	Once for each pair of sites being reported.	The average duration of all the media streams used between the two sites. The average is total duration of the media streams between the two sites divided by the number of such media streams.

B.5.10 WAN Media Stream Detail Report

The WAN Media Stream Detail Report (Figure B-14) shows details of each media stream placed over the WAN. You can configure the report to display information for all calls or for only intersite calls.

Media Stream Detail Report

Starting: 12:00:00AM Ending: 11:59:59PM All Calls

Site A	Site B	Start Time	WAN	Call ID	Encoding	Max Jitter (ms)	% Packets Lost	Duration
Austin	Austin	6/30/2010 6:09:01PM	N-NS	FFD19672	G722	7	0.42	00:00:14
Austin	Austin	6/30/2010 7:40:29PM	N-NS	FACDBD23	G722	0	0.04	00:01:31
Austin	Austin	6/30/2010 7:42:43PM	N-NS	E6BE57E7	G722	0	0.00	00:00:22
Austin	Austin	6/30/2010 7:44:08PM	N-NS	141CB0F1	G722	0	0.00	00:00:58
Austin	Austin	6/30/2010 7:46:40PM	N-NS	05112504	G722	0	0.00	00:00:45
Austin	Austin	7/6/2010 1:09:12PM	N-NS	57EA2B63	G722	5	1.83	00:00:03
Austin	Austin	7/6/2010 1:11:33PM	N-NS	39D31389	G722	2	0.00	00:00:01
Austin	Austin	7/6/2010 2:03:30PM	N-NS	8790D62D	G722	0	0.00	00:00:18
Austin	Austin	7/6/2010 2:04:25PM	N-NS	7D89238F	G722	0	0.00	00:00:20
Austin	Austin	7/7/2010 2:18:58PM	N-NS	8044AB1E	LINEAR	0	0.00	00:00:00
Austin	Austin	7/8/2010 10:32:43AM	N-NS	E4CD5A8D	G722	0	0.00	00:01:04
Austin	Austin	7/9/2010 10:07:07PM	N-NS	4714C6BA	G722	11	0.00	00:00:01
Austin	Austin	7/9/2010 10:07:16PM	N-NS	4210ED6B	G722	2	0.00	00:00:01
Austin	Austin	7/19/2010 9:20:30PM	N-NS	A87B472D	LINEAR	10	0.00	00:00:08
Austin	Austin	7/20/2010 3:35:38PM	N-NS	6D7E491B	G722	0	0.00	00:01:05

Austin-Austin Total **15 Call(s) - Average Duration 00:00:27 - Total Duration 00:06:51**

Figure B-14 WAN Media Stream Detail Report

B.5.10.1 Call Included

See the Media Stream Summary Report for information about selection. This report calls out each media stream established between two sites.

IP phone media streams are not included in this report.

The Media Stream Detail Report is described in Table B-13.

Table B-13 WAN Media Stream Detail Report Field Descriptions

Field	Presence/ Frequency	Description
Site A	Once for each media stream.	The name of the site. This data is retrieved from the ASiteName field in the MediaStream table.
Site B	Once for each media stream.	The name of the site that communicates to Site A. This data is retrieved from the BSiteName field in the MediaStream table.
Start Time	Once for each media stream.	The time the media stream began. This data is retrieved from the StartTime field in the MediaStream table.
WAN	Once for each media stream	"Yes" indicates the media stream accessed the WAN. "No" indicates the media stream did not access the WAN.
CallID	Once for each media stream.	The Call Identification number for the media stream listed on the detail report. By matching the CallID in the report to the CallID of a WAN call with voice quality issues, you can understand the cause of the problems. This data is retrieved from the CallID field in the MediaStream table.

Table B-13 WAN Media Stream Detail Report Field Descriptions (Continued)

Field	Presence/ Frequency	Description
Encoding	Once for each media stream.	The method of voice encoding used for the media stream. This data is retrieved from the EncodingType field in the MediaStream table. The EncodingType field can have the following values for encoding methods: 1 ALAW – PCMA/8000 (or G711A) 2 MULAWPCMU/8000 (or G711µ) 3 LINEARL16/16000 4 ADPCMDVI4/8000 5 G729AG729A/8000 6 G729BG729B/8000 7 LINEARWIDEBANDL16/16000 8 G722G722/8000 9 BV32BV32/16000 10 BV16BV16/8000 11 AAC_LC32000AAC_LC/32000 12 CustomCodec added by administrator
Max Jitter	Once for each media stream.	The maximum jitter encountered. This value is the maximum of the A MaxJitter or B MaxJitter for the record in the MediaStream table for the media stream. If a significant number of calls are reported with a Max Jitter value close or equal to the Maximum Jitter Buffer value, you may want to increase the Maximum Jitter Buffer or investigate the cause of excess jitter in the network.
% Packet Loss	Once for each media stream.	This is the number of packets that were expected to arrive but did not arrive at the destination. Lost packets were mostly likely dropped on their way through the network.
Duration	Once for each media stream.	The duration of time the media stream was used across the WAN connection. This data is retrieve from the DurationSeconds field of the MediaStream table record for this call.

B.5.11 Account Code Summary Report

Summarizes call information for each account, including number of calls each day, along with their total and average duration. There are also totals for the reporting period. This report allows the administrator to indicate whether there should be summary information for each account. If this is desired, each extension's use of the account is summarized. If not, there's a simple total for the entire account code.

Account Codes are only applicable to outgoing calls. Outgoing calls are identified in the Call table of the CDR database with the CallType field set to 3 (Outbound). The only outgoing calls that appear in the report are those calls for which an Account Code was collected (the account code is recorded in the BillingCode field of the Call table). If the user does not provide an account code on an outgoing call (because it isn't required, or it is optional and they choose not to provide it), that call does not show up on the report. Figure B-15 is an example of the Account Code Summary Report.

Account Code Summary Report

Start: End:

Start: 12:00:00AM End: 11:59:59PM

		Total Calls	Total Duration	Average Duration
10001 (AAcct1)				
	07/09/2010	2	00:09:08	00:04:34
	07/19/2010	3	00:00:53	00:00:17
	Total	**5**	**00:10:01**	**00:02:00**
10003 (CAcct3)				
	07/19/2010	1	00:00:16	00:00:16
	Total	**1**	**00:00:16**	**00:00:16**
	Grand Total	**6**	**00:10:17**	**00:01:42**

Figure B-15 Account Code Summary Report

The Account Code Summary Report fields are described Table B-14.

Table B-14 Account Code Summary Report Field Descriptions

Field	Presence/Frequency	Description
Account	Shown once for each account being reported on.	The account code that users enter to identify the account that a call is logged against. The name of the account code as configured in Director is also shown. For example, “300 (Marketing),” where “300” is the account code and “Marketing” is the name. For each call where account code information is collected, the account code is stored in the BillingCode field of the Call table. The name of the account code is stored in the FriendlyBillingCode field.
User	Only shown if the report option is selected to “Enable User Breakdown.” Shown once for each user who made calls for the account being reported upon.	The name and extension of the user who originated calls summarized in the report. For example, “John Smith (x3415).” The Extension field in the Call table identifies the party who originated the account code call. The extension field is where the report gets the extension number information. The name of the user comes from the PartyID (the first name) and PartyIDLastName (the last name) fields of the Connect record for the party that originated the call. The Connect record is tied to the Call table record, by the CallTableID field in the Connect table. All the Connect records for a particular call have the same CallTableID setting.

Table B-14 Account Code Summary Report Field Descriptions (Continued)

Field	Presence/Frequency	Description
Total Calls	Repeated for each row.	The total number of calls for a particular day for the account. The total is broken down by user within each account if the “Enable User Breakdown” option is selected. A call is reported for the day on which the call started. That is, if a call starts on one day but ends on the next day, it is only reported for the day on which it started. The start of the call comes from the StartTime field in the Call table record for each call.
Total Duration	Repeated for each row.	The total duration of the calls being reported on the row. The duration is the total call duration, even if the call was transferred to parties such that the originator of the call was not on the call for the entire period. Duration is reported in the day the call started, but includes the entire call duration. For example, a call starts on 1/17 (with 20 minutes on 1/17) and ends on 1/18 (with 30 minutes on 1/18). The call is reported on 1/17 with duration of 50 minutes. This is then included in the total duration for all calls on 1/17. The duration of each call in this report comes from the Duration field in the call table.
Average Duration	Repeated for each row.	Calculated by dividing the total duration for a row by total calls.

B.5.12 Account Code Detail Report

The Account Code Detail Report (Figure B-16) provides a detailed list of calls that occurred for each account. For each call the date and time of the call, number dialed, the extension making the call, and the duration of the call is included. For each account, a summary is provided of the number of calls, along with their total and average duration.

Account codes are only applicable to outgoing calls. Outgoing calls are identified in the Call table of the CDR database with the CallType field set to 3 (Outbound). The only outgoing calls that appear in the report are those calls for which an account code was collected (the account code is recorded in the BillingCode field of the Call table). If the user does not provide an account code on an outgoing call (because it isn't required, or it is optional and they choose not to provide it) that call does not show up on the report.

Account Code Detail Report

Start:					**End:**
Start: 12:00:00AM					**End: 11:59:59PM**
	Date	**Time**	**Dialed Number**	**Calling Extension**	**Duration**
10001 (AAcct1)					
	7/9/2010	7:53:33PM	+14083313609	126	00:04:29
	7/9/2010	7:54:52PM	+14083313609	122	00:04:39
	7/19/2010	8:42:30PM	+14083313609	131	00:00:10
	7/19/2010	8:43:44PM	+14083313609	131	00:00:09
	7/19/2010	8:53:46PM	+14083313609	131	00:00:34
Total	**5 Call(s)**	**Total Duration: 00:10:01**	**Average Duration: 00:02:00**		
10003 (CAcct3)					
	7/19/2010	8:44:16PM	+14083313609	131	00:00:16
Total	**1 Call(s)**	**Total Duration: 00:00:16**	**Average Duration: 00:00:16**		
Grand Total			**6 Call(s)**	**00:10:17**	**Total 00:01:42 Average**

Figure B-16 Account Code Detail Report

The Account Code Detail Report fields are described in Table B-15.

Table B-15 Account Code Detail Report Field Descriptions

Field	Presence/ Frequency	Description
Account	Shown once for each account being reported on	The account code that users enter to identify the account that a call is logged against. The name of the account code as configured in Director is also shown. For example, “300 (Marketing),” where “300” is the account code and “Marketing” is the name. For each call where account code information is collected, the account code is stored in the BillingCode field of the Call table. The name of the account code is stored in the FriendlyBillingCode field.
Date	Repeated for each call.	The date on which the call started. A call is reported for the day on which the call started. That is, if a call starts on one day, but ends on the next day, it is only reported for the day that it started on. The Date is extracted from the StartTime field in the Call record for each call in the report.
Time	Repeated for each call.	The time at which the call started. Time comes from the StartTime field in the Call table record for each call in the report.

Table B-15 Account Code Detail Report Field Descriptions (Continued)

Field	Presence/ Frequency	Description
Dialed Number	Repeated for each call.	The number that was dialed to begin the call. Dialed Number comes from the DialedNumber field in the Call table record for each call in the report.
Calling Extension	Repeated for each call.	The number of the user that originated the call. Calling Extension comes from the Extension field in the Call table record for each call in the report.
Duration	Repeated for each call.	The total duration of the calls being reported on the row. The duration is the call duration, even if the call was transferred to parties such that the originator of the call was not on the call for the entire period. The duration of each call in this report comes from the Duration field in the Call table.

B.6 CDR Database

This appendix specifies how data is stored in the CDR database tables. The CDR database records call data in the following tables:

- **Call Table:** An entry is made in the Call table for each call in the ShoreTel system. Other tables often reference the entries to the Call table.
- **Connect Table:** An entry is made in the Connect table for each connection to a call. When used with the Call table, a complete call history is provided.
- **MediaStream Table:** An entry is made in the MediaStream table each time there is a media stream between two switches that are at different sites. In some cases, such as for conference calls, there may be multiple media streams per call.
- **AgentActivity Table:** An entry is made in the AgentActivity table each time a workgroup agent logs into a workgroup and when he or she completes wrap-up.
- **QueueCall Table:** An entry is made in the QueueCall table for each call that is handled by a workgroup server. The entry identifies how the call leaves the workgroup—either by abandonment or for handling.
- **QueueStep Table:** An entry is made in the QueueStep table for each step where the workgroup server either hunts for agents or walks through workgroup queue steps. This provides more detailed information about how the call was disposed of by the workgroup server.
- **QueueDepth Table:** An entry is made in the QueueDepth table each time the depth of a workgroup server's call queue changes.

In addition to the tables listed above, the database contains a number of enumeration tables, which are documented below when discussing the tables that reference these enumeration/lookup tables.

Logged data is reflective of the time of the logging. For example, certain records contain the name of a trunk group from the configuration database. The name of the trunk group may be changed in the configuration database. New log entries reflect the changed name, but existing logs continue to have the old name.

B.6.1 Call Table

The CDR database reflects all calls within the system with a few exceptions which are listed below. These exceptions reflect the ShoreTel Telephony Management Server (TMS) that allows calls to continue even when portions of the system or network are not available. As the TAPI service provider for the ShoreTel Server, TMS manages the call control communications between all other ShoreTel services.

The exceptions are:

- If TMS is not connected to any of the call endpoints, the call is not recorded in the Call table. Because of network outages, TMS may not be connected to call endpoints, yet the call endpoints may have the connectivity necessary to complete the call (for example, the switches are able to communicate with each other but not to TMS).
- If TMS is not connected to some of the call endpoints (for example, a switch involved in the call), the information about the call can be incomplete (for example, the information in the Connect table as explained in the next section would only reflect some of the parties involved in the call).
- If TMS is restarted, any call entries that were incomplete, along with their associated Connect entries are destroyed. Incomplete calls do not show “Yes” in the locked field.
- Also at TMS restart time, TMS logs any calls in progress.

Figure B-17 illustrates how new entries are added to the Call table whenever there is a call in the ShoreTel system. Note that an entry is added to the Call table when the call begins (or when TMS starts up, for any calls in progress) and is updated when the call ends.

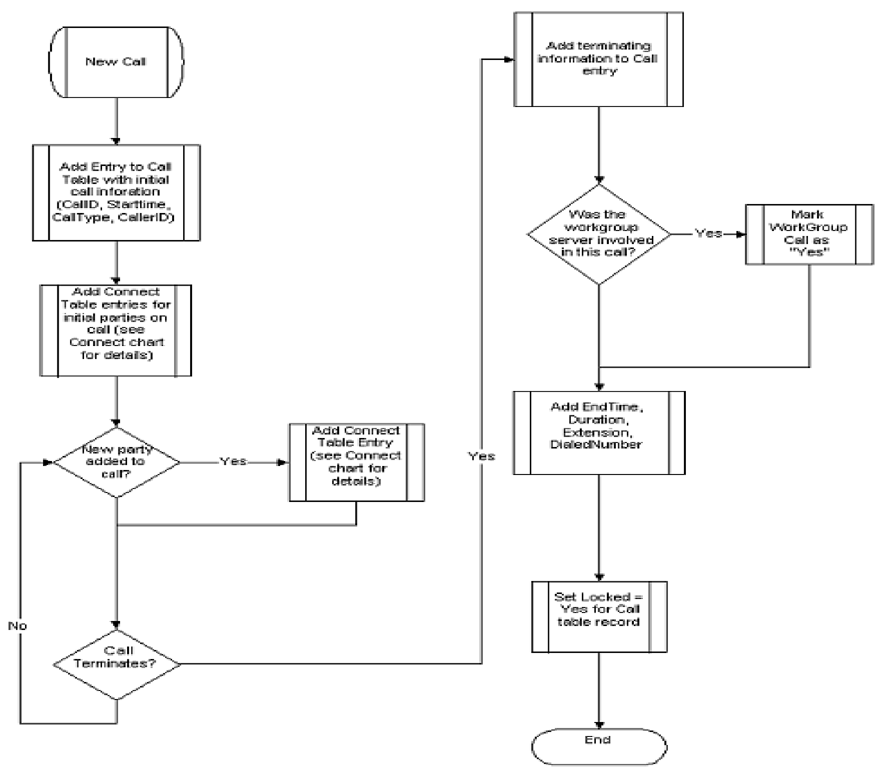


Figure B-17 New Entries in the Call Table

The Call table is reference by other tables, most important among them being the Connect table. You can analyze the Call and Connect tables to understand the complete disposition of a call as attempts are made to add parties, transfers occur, and so on. Other tables can index the Call table, through the primary key “ID,” which is unique for each record.

There is a CallID field that is used internally by the ShoreTel system to identify calls. This, however, should not be used as the index into the table).

Close examination of the Call table shows that there are more calls recorded than you may initially expect. For example, if a call is made to a workgroup, you will see an initial call, generally from an incoming trunk. As agents are hunted, calls are made by the workgroup server to agents. If multiple agents are hunted, there will be multiple calls. Once one of the agents is successfully hunted, if you looked at the Connect table you see the agent being attached to the original call. Table B-16 provides information about the elements in the Call Table.

Table B-16 Call Table Field Descriptions

Field Name	Data Type	Description
ID	AutoNumber	Unique identifier. (4-byte integer, required)
CallID	Number	Number for the existence of the call. (4-byte integer)
SIP GUID	Text	SIP Global ID number (32 characters, zero-length)
StartTime	Date/Time	For an inbound call, this is when the trunk has been seized. For an outbound call, this is when the user has completed dialing. (8-byte date/time, required)
StartTimeMS	Number	Append this information to the StartTime to reduce the absolute start time to the millisecond when the call began. (2-byte integer, required)
EndTime	Date/Time	Time when the call terminates (either by the near end hanging up or when the end external to the system hangs up) and the ShoreTel switch receives the notification of the disconnect. (8-byte date/time)
EndTimeMS	Number	Append this information to the StartTime to reduce the absolute start time to the milliseconds of when the call began (milliseconds). (2-byte integer, required)
CallNote	Text	User entered Call Note. This can be added from the ShoreTel desktop client. (64 characters, zero-length)
BillingCode	Text	Account code assigned to the call. (32 characters, zero-length)
Locked	Yes/No	Read-only status for this call (set once call has ended). Not locked means the call is still in progress. (boolean)
Extension	Text	For an outbound or extension-to-extension call, the extension has the dialed number of the originator of the call. This field is blank for an outbound call from an anonymous phone with no currently assigned DN. For an inbound call, the extension field contains the DN of the last party involved in the call (excluding voice mail or auto-attendant). For instance, an incoming call to an extension that transferred the call to extension 300 has “300” in the extension field (the complete history of parties connecting to the call can be found on the Connect table). All calls to an extension that forwarded to voice mail have the extension of the called party and not the voice mail number. (15 characters, zero-length)

Table B-16 Call Table Field Descriptions

Field Name	Data Type	Description
Duration	Date/Time	Elapsed time of the call from beginning to end. Calculated by subtracting StartTime from EndTime. Start time begins when the first party is added to a call. End time is when the last party leaves resulting in the end of the call. (8-byte date/time).
CallType	Number	See enumeration in CallType table. (1-byte integer, required)
WorkGroupCall	Yes/No	Is this a workgroup call? Yes indicates that the workgroup server was involved in processing the call. If the call was directed toward a workgroup server, but that server was unavailable, then this field is set to "No" because the workgroup server never becomes involved in the call. (boolean)
LongDistance	Yes/No	From the perspective of the trunk for the call, did this call involve a long distance connection? The first connect record of the call is used to determine whether a call is long distance. If the first leg is an extension call, the value is always No. A trunk call can be transferred or conferenced, so the total long distance time can only be determined by examining all Connect records. (boolean)
DialedNumber	Text	Extension-to-extension and outbound: Number dialed plus trunk access code if any. (15 characters, zero-length)
CallerID	Text	For CallType=Inbound only: Caller-ID number if present. If blocked or unavailable text is provided by the PSTN to indicate caller ID as unavailable it is included here; for example, the text may be "blocked" or "unavailable" (15 characters, zero-length)
Archived	Yes/No	Has this call been archived? (boolean)

B.6.2 Enumeration Tables: Use for the Call Table

B.6.2.1 Call Type

Table B-17 Call Type Descriptions

Call Type	Name	Description
0	Null	
1	ExtToExt	Extension-to-extension call.
2	Inbound	A trunk is the originating party.
3	Outbound	An extension is originating and a trunk is called.

B.7 Connect Table

The Connect table contains a record for each party in a call. There are many different types of parties that can be reflected in the table including individual user extensions, workgroups, workgroup agents, and trunks.

Figure B-18 illustrates how new entries are added to the Connect table each time a party is added to a call within the ShoreTel system.

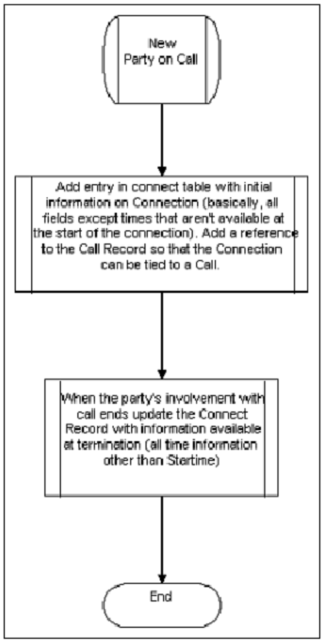


Figure B-18 New Entries in the Connect Table

B.7.1 Connect Table Field Descriptions

Table B-18 describes the Connect Table data fields.

Table B-18 Connect Table Field Descriptions

Field Name	Data Type	Description
ID	Auto-Number	Unique identifier. (4-byte integer – required)
PartyType	Number	Party that initiated the call. Value corresponds to value from Connect Type Party Type table (Table B-19). (6-bit integer – required)
CallTableID	Number	Link to Call Table ID Key. (20-bit integer – required)
LineID	Number	TAPI permanent line ID for this party. (20-bit integer – required)
SwitchID	Number	Party's Switch ID – unique ID assigned to configuration database to ShoreTel switch. Data only available through database – not Director. Value of 0 indicates the party is a workgroup, voicemail, or an unassigned user. In these cases, DN is are not assigned to a switch/ port. (11-bit integer)
PortNumber	Number	Party's port number corresponding to physical port or channel number on the ShoreTel switch. Value of 0 indicates the party is a a workgroup, voicemail or unassigned user. (11-bit integer)

Table B-18 Connect Table Field Descriptions (Continued)

Field Name	Data Type	Description
PortID	Number	Party's port ID – if any. Unique ID assigned to ShoreTel switch port by configuration database. Data available through database – not Director. Value of 0 indicates the party is a workgroup, voicemail, or an unassigned user. (11-bit integer)
PortName	Text	Name of port (Trunk or Extension) – user defined in Director. (50 characters)
GroupID	Number	Unique ID assigned by configuration database. (11-bit integer) Value is a TrunkGroupID if PartyType=Trunk. Value is UserGroupID if PartyType=Station. Value is 0 if it is not applicable.
GroupName	Number	Data is defined is Director by use. (50 characters) If PartyType=Station, value is name of the User-Group If PartyType=Trunk, value is Trunk-Group.
ConnectTime	Date/ Time	Time when party was added to call. Initial parties on inbound call, value indicates time trunk was seized. Initial parties on outbound call, value indicates time dialing was complete.
ConnectTimeMS	Number	Append to ConnectTime to determine start time of call with millisecond precision. (11-bit integer, milliseconds)
DisconnectTime	Date/ Time	Time when party disconnected from call.
DisconnectTimeMS	Number	Append to DisconnectTime to determine end time of call with millisecond precision. (11-bit integer, milliseconds)
ConnectReason	Number	Connect reason code. Refer to Table B-21. (6-bit integer – required)
DisconnectReason	Number	Disconnect reason code. Refer to Table B-22. (6-bit integer – required)
PartyIDFlags	Number	Caller ID flags that specifies the data available in ID and Name fields Internal party: number, name and last name from system address book. External party: data corresponds to caller ID field provided by the PSTN. Refer to Table B-20. (6-bit integer – required)
PartyID	Text	Number of party. Refer to PartyIDFlags field. (50 characters)
PartyIDName	Text	Name of party. Refer to PartyIDFlags field. (50 characters)
PartyIDLastName	Text	Last name of party. (50 characters) Field is blank for external party – PartyIDName contains first and last name, as provided by PSTN Caller ID service.
CtrlPartyIDFlags	Number	Caller ID flags that specifies the data available in ID and Name fields for the controlling party. Controlling party causes the event. Example: for an entry listing a call was transferred from extension 400 to extension 300, the controlling party is extension 400. Original call will not have a control party. Refer to Table B-20. (6-bit integer – required)

Table B-18 Connect Table Field Descriptions (Continued)

Field Name	Data Type	Description
CtrlPartyID	Text	Number of controlling party. Refer to CtrlPartyIDFlags field. (50 characters)
CtrlPartyIDName	Text	Name of controlling party. Refer to CtrlPartyIDFlags field. (50 characters)
CtrlPartyIDLastName	Text	Last name of controlling party. (50 characters)
MailboxID	Text	Mailbox ID if PartyType=VMForward or VMLogin PartyType=VMForward – specifies mailbox receiving forwarded message. PartyType=VMLogin – specifies original target mailbox.
RelatedCallTableID	Number	Reserved. (20-bit integer).
TalkTime	Date/ Time	Total connect time. Calls with more than 24 hours include the date. Date not included on calls shorter than one hour. (8-byte date/ time) Example: A 25 hour call has a TalkTime of 1 day and 1 hour.
TalkTimeSeconds	Number	The seconds component of the TalkTime. (20-bit integer, seconds)
HoldTime	Date/ Time	Time on hold. Includes date on calls with more than 24 hours hold time. Date not included on calls with less than one hour hold time. (8-byte date/time) Example: A call with 25 hour time has a HoldTime of 1 day and 1 hour.
RingTime	Date/ Time	Inbound calls: time spent offering Outbound calls: ringback time.
Duration	Date/ Time	The time between ConnectTime and DisconnectTime
LongDistance	Number	Lists trunk connected long distance for outbound calls if PartyType=trunks
TrunkDirection	Number	Indicates inbound / outbound direction. Refer to TrunkDirection Enumeration Table.
SecurityFlag	Number	AES/SRTP flags that indicates call encryption status. (6-bit integer)
SiteName	Text	Reserved (50 characters)
ServerName	Text	Reserved (64 characters)

B.7.2 Enumeration Tables Used for Connect Table

B.7.2.1 PartyType

Table B-19 lists the Connect Table party types.

Table B-19 Party Type Enumeration Table

Type #	Party Type	Description
0	Null	
1	Station	User extension which is currently assigned a home port; sometimes referred to as a “logged in user”.
2	Trunk	Trunk (of any kind).
3	Virtual	A user extension which does not currently have an assigned home port (sometimes referred to as a “logged out user”).
4	Workgroup	Workgroup extension.
5	AutoAttendant	Auto-Attendant extension.
6	VMForward	Voice mail forward extension (take a message).
7	VMLogin	Voice mail login extension.
8	BackupAA	Backup auto-attendant (built into switch).
9	AnonPhone	Anonymous telephone.
10	Nightbell	Nightbell extension.
11	Paging	Paging extension.
12	WorkgroupAgent	Records marked as WorkgroupAgent for calls transferred from a Workgroup to an Agent. Direct inbound calls to an agent are Station type.
13	Unknown	Unknown type.
14	RoutePoint	Route point.

B.7.2.2 PartyIDFlag

Table B-20 lists the Connect Table party ID flags.

Table B-20 Party ID Flag Enumeration Table

Flag #	Party ID Flag Name	Description
0	Null	
1	Blocked	Blocked
2	OutOfArea	Out-Of-Area
4	Name	Name
8	Address	Address
12	NameAddress	Name & Address
16	Partial	Partial
32	Unknown	Unknown
64	Unavailable	Unavailable

B.7.2.3 ConnectReason

Table B-21 lists the Connect Table connect reason codes.

Table B-21 Connect Reason Enumeration Table

Connect #	Connect Reason	Description
0	Null	
1	Direct	TMS was not available when the party connected to the call. Connection information is logged, but there is no more ConnectReason information.
2	ForwardBusy	The party was connected because the previous party's call handling mode was set to forward calls if the previous party was busy.
3	ForwardNoAnswer	The party was connected because previous party's call handling mode was set to forward calls if the previous party didn't answer.
4	ForwardAll	The party was connected because previous party's call handling mode was set to forward all calls.
5	Pickup	The call was connected because the called party answered the call.
6	Unpark	Unpark
7	Redirect	Redirect
8	Completion	Completion
9	Transfer	The call was connected after the call was transferred to the party.
10	Reminder	Reminder
11	Unknown	Unknown
12	Unavailable	Unavailable
13	Intrude	Intrude
14	Parked	Parked
15	CampedOn	CampedOn
16	RouteRequest	RouteRequest
17	Called	The party was added to the call because it was the initial target of the call.
18	Forward	Forward
19	Originate	The party initiated this call.
20	Conference	The party was added to the call because the party was conferenced into the call.

B.7.2.4 Disconnect Reasons

Table B-22 lists the Connect Table disconnect reason codes.

Table B-22 Disconnect Reason Enumeration Table

Reason #	Disconnect Reason	Description
0	Null	
1	Normal	Normal termination
2	Unknown	Unknown reason
3	Reject Call	Call was rejected
4	Pickup Call	Call picked up by other destination
5	Forwarded Call	Call forwarded to another destination
6	Busy	Busy destination
7	NoAnswer	No answer by destination
8	BadAddress	Bad address
9	Unreachable	Destination cannot be reached
10	Congestion	Inadequate bandwidth
11	Incompatible	Destination is incompatible
12	Unavailable	Destination is unavailable
13	NoDialTone	No dial tone from the trunk
14	NumberChanged	Destination number changed
15	OutOfOrder	Destination out of order
16	TempFailure	Temporary failure
17	QoSUnavailable	QoS not available
18	Blocked	Destination blocked
19	DoNotDisturb	Do not disturb
20	Cancelled	Call cancelled
21	Unpark	Call unparked to different destination

B.7.2.5 Trunk Direction

Table B-23 lists the Trunk Direction flags.

Table B-23 Trunk Direction Enumeration Table

Flag #	Party ID Flag Name	Description
2	Inbound	The trunk direction is established by the central office
3	Outbound	The trunk direction is established by the local system.

B.8 MediaStream Table

The MediaStream table logs media information about InterSite Calls. At a high level, there is one such entry for each InterSite call. Information about both parties involved in the call is recorded. Table B-24 describes the elements in the MediaStream Table.

Table B-24 Media Stream Data Field Descriptions

Field Name	Data Type	Description
ID	AutoNumber	Unique identifier. (4-byte integer, required)
CallID	Number	Unique number for the existence of the call. (4-byte integer)
SIP GUID	Text	SIP Unique Global ID number (32 characters, zero-length)
EncodingType	Number	Encoding type used for media stream. (1-byte integer)
PayloadSize	Number	Media payload size in bytes for each media packet. (4-byte integer)
StartTime	Date/Time	Date and time the call started.
Duration	Date/Time	Elapsed time of call from begin to end. (8-byte date/time)
DurationSeconds	Number	Elapsed seconds time of call from begin to end. (4-byte integer)
FailureCode	Number	Error code. See MediaFailureCode table for enumeration. (1-byte integer)
A PartyType	Number	Party A's type enumeration. See the PartyType table. (1-byte integer)
A SiteID	Number	Party A's Site ID. (4-byte integer)
A SiteName	Text	Party A's Site Name. (50 characters, zero-length)
A LineID	Number	TAPI permanent line ID for party A. (4-byte integer)
A Name	Text	Call type name for party A. (50 characters, zero-length)
A_Extension	Text	Call extension number for party A (32 characters, zero-length)
A IP Address	Text	Local IP Address for party A. (15 characters, zero-length)
A TotalPackets	Number	Total packets received by party A. (4-byte integer)
A LostPackets	Number	Total packets lost by party A. (4-byte integer)
A MaxJitter	Number	Maximum jitter (ms) for party A. (4-byte integer)
A Underruns	Number	Number of receive underruns for party A. (4-byte integer)
A Overruns	Number	Number of receive underruns for party A. (4-byte integer)
B PartyType	Number	Party B's type enumeration. (1-byte integer)
B SiteID	Number	Party B's Site ID. (4-byte integer)
B SiteName	Text	Party B's Site Name. (50 characters, zero-length)
B LineID	Number	TAPI permanent line ID for party B. (4-byte integer)
B Name	Text	Call type name for party B. (50 characters, zero-length)

Table B-24 Media Stream Data Field Descriptions (Continued)

Field Name	Data Type	Description
B_Extension	Text	Call extension number for party B (32 characters, zero-length)
B IP Address	Text	Local IP Address for party B. (15 characters, zero-length)
B TotalPackets	Number	Total packets received by party B. (4-byte integer)
B LostPackets	Number	Total packets lost by party B. (4-byte integer)
B MaxJitter	Number	Maximum jitter (ms) for party B. (4-byte integer)
B Underruns	Number	Number of receive underruns for party B. (4-byte integer)
B Overruns	Number	Number of receive overruns for party B. (4-byte integer)
InterSite	Yes/No	Indicates a logged call is InterSite. Only Intersite calls are logged.
Archived	Yes/No	Has this entry been archived? (boolean)

B.9 AgentActivity Table

The AgentActivity Table has information about the workgroup agents' availability. Entries are made to record agents' Login/Logout from the workgroup and to reflect their time in Wrapup mode. Figure B-19 illustrates the flow of new entries being added to the AgentActivity table.

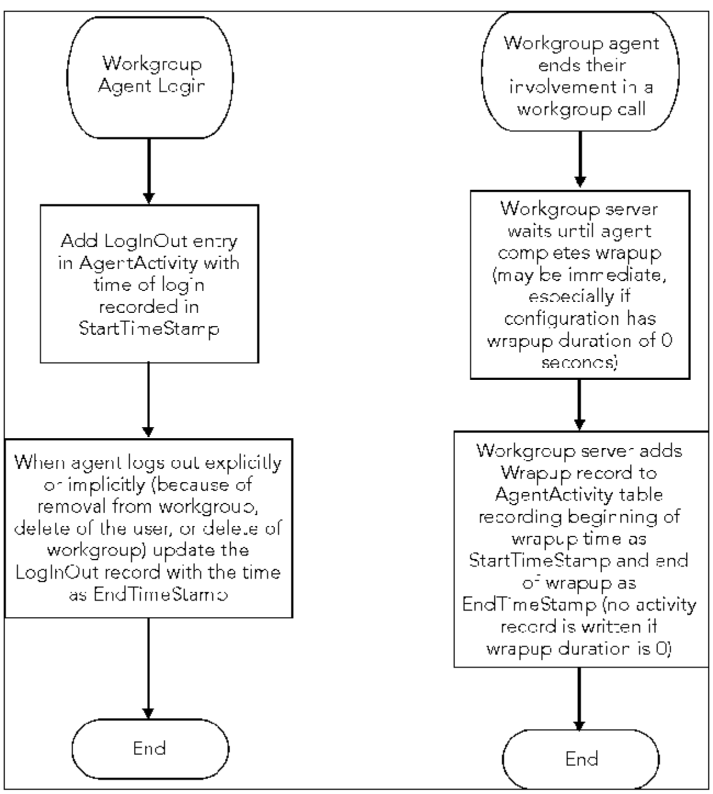


Figure B-19 Entries to the AgentActivity Table

The left flow shows how each time a workgroup agent logs in, a LogInOut entry is added, which is then updated at logout time. The right flow shows how the AgentActivity table is also updated as agents complete their handling of workgroup calls. Table B-25 describes the elements in the Agent Activity table.

Table B-25 Agent Activity Table field descriptions

Field Name	Data Type	Description
ID	AutoNumber	Unique identifier. (4-byte integer, required)
CallID	Number	Unique number for the existence of the call. Provided in wrapup records. (4-byte integer)
AgentDN	Text	WorkGroup Agent's dialed number (extension). (15 characters, zero-length)
AgentFirstName	Text	WorkGroup Agent's First Name (50 Characters, zero-length).
AgentLastName	Text	WorkGroup Agent's Last Name (50 Characters, zero-length) (may be blank if the agent doesn't have a last name in the configuration database)
State	Number	Enumerated Agent State—set AgentStateLUT for possible values.
WorkGroupDN	Text	WorkGroup dialed number (extension) for which this agent activity is for (15 characters, zero-length)
WorkGroupName	Text	Workgroup's name. (50 Characters, zero-length)
StartTimeStamp	Date/Time	Start time stamp. For LogInOut records, StartTimeStamp indicates the time when the agent logged into the workgroup. For wrapup records, the StartTimeStamp indicates the time when the agent entered wrapup time. See notes below. (8-byte date/time).
EndTimeStamp	Date/Time	End time stamp (8-byte date/time).
Archived	Yes/No	Has this entry been archived? (boolean)

- Two types of records are placed in the AgentActivity table. The State field identifies the type of record.
- LogInOut Records record the time that an agent is logged into the workgroup.
- Wrapup records record the time that an agent is in wrapup state.
- All records in the table should have ID, AgentDN, AgentFirstName, AgentLastName (unless blank), State, WorkGroupDN, WorkGroupName, StartTimeStamp, and Archived.
- LogInOut Records may exist for agents that have Logged into the workgroup but have not yet logged out. For these records the StartTimeStamp indicates the time when the agent logged into the workgroup. The EndTimeStamp is updated when the agent logs out of the workgroup with the time of the logout.
- For wrapup records the StartTimeStamp indicates the time when the agent entered wrapup time and EndTimeStamp indicates when they exit wrapup state.

- Wrapup records can contain a CallID to identify the Call that the agent was wrapping up from for the Wrapup record. This will not be provided in cases where the agent is manually placed in wrapup state when not on a call.
- There is always a wrapup record when an agent wraps up a call, even for the case where wrapup time is set to zero.

B.9.1 Enumeration Tables Used for AgentActivity

Table B-26 lists the Agent Activity Enumeration Tables.

Table B-26 Agent Activity Enumeration Table

State #	AgentState	Description
0	Null	
1	Reserved	Previously used for Login
2	Reserved	Previously used for Logout
3	Wrap_Up	Agent performing post-call wrap-up
4	Reserved	Temporarily used for Outcalls
5	LogInOut	Agent Login later updated with Logout time.
6	SecLogInOut	Secondary login activity for agents belonging to multiple Workgroup.

Login and Logout are no longer used.

B.10 QueueCall Table

Each time a call is processed by the workgroup server, an entry is made in the QueueCall table. A workgroup is a call queuing mechanism, thus the name “QueueCall” in the CDR database.

Each time a call is made to a workgroup when the workgroup server is operational, an entry is made in the QueueCall table; moreover, there is only one entry for each call. In other words, one and only one entry appears for each call. A call can be made to the workgroup dialed number, but if the workgroup server does not process the call, an entry is not made in the QueueCall table for the call. Moreover, the call will not be marked as a workgroup call in the call table.

Figure B-20 illustrates how updates are made to the QueueCall table.

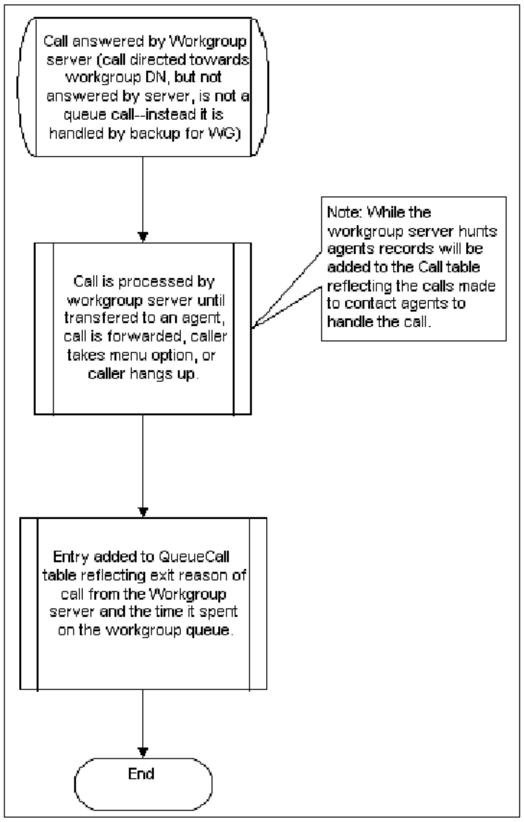


Figure B-20 New Entries to the CallQueue Table

Each entry in the QueueCall table contains the following fields as shown in Table B-27.

Table B-27 Queue Call Table Field Descriptions

Field Name	Data Type	Description
ID	AutoNumber	Unique identifier. (4-byte integer, required)
CallID	Number	Unique number for the existence of the call. (4-byte integer)
ConnectTableID	Number	Link to Connect Table ID Key. You can find more information about the connection to the call in the connect table. The Connect table entry here is for the Workgroup DNs connection to the call. If you want information from the Call table entry for this call, the reference to the Call table in the Connect entry should be used to find the Call table entry. (4-byte integer)
StartTime	Date/Time	The time at which the call is answered by the workgroup server, thereby beginning it's time on the call queue (workgroup) DN. (8-byte date/time)
Duration	Date/Time	Time from when the call is offered to the workgroup DN until it leaves the call queue. The call leaves the queue when it is answered by an agent, is abandoned by the calling party, or leaves the queue for other reasons. The complete lists of reasons for leaving the queue are found in the QueueExitReasonLUT table. (8-byte date/time)
DurationSeconds	Number	Duration expressed in number of seconds. (4-byte integer)

Table B-27 Queue Call Table Field Descriptions

Field Name	Data Type	Description
QueueName	Text	Name of the call queue (workgroup). (50 characters, zero-length)
QueueDN	Text	Extension number of the call queue (workgroup). (15 characters, zero-length)
ExitReason	Number	Enumerated reason the call left the call queue (see the QueueExitReasonLUT for enumerations). (1-byte integer)
TargetType	Number	Enumerated type of handoff target (see TargetTypeLUT for enumerations). (1-byte integer)
TargetFirstName	Text	Name or first name of target. (50 characters, zero-length)
TargetLastName	Text	Last name of target if applicable (blank if the target agent doesn't have a last name in the configuration database). (50 characters, zero-length)
TargetDN	Text	Dialed number of target. (15 characters, zero-length)
Archived	Yes/No	Has this entry been archived? (boolean)

- Partial records are never written. A record is written only once, either when the call is abandoned, the call is connected to an agent, or leaves the queue for other reasons as enumerated in QueueExitReasonLUT.
- If QueueExitReason = Abandon, target information (TargetType, TargetFirstName, TargetLastName, TargetDN) is meaningless and will be blank.
- If QueueExitReason is TransferToAgent, the TargetFirstName and TargetLastName are filled in with information about the agent.
- If the QueueExitReason is Forwarding (2, 3, 4, or 5 for forward always, busy, no answer, or no logged in agent) or transfer (9, 10, and 11 for transfer to a menu, extension or voice mail), the DN that the call is being forwarded or transferred to is provided in the TargetDN field. However, the TargetFirstName and TargetLastName are not provided.
- A QueueExitReason is always entered. The field will never be blank. "Unknown" will only be used in the case of failure (and maybe not at all).

B.10.1 Enumeration Tables Used for QueueCall

Table B-28 describes the elements in the Queue Call Exit Reason Enumeration table.

Table B-28 Queue Call Exit Reason Enumeration Table

ExitReason	Name	Description
0	Null	
1	TransferToAgent	Hunt succeeded and transferred to agent.
2	ForwardAlways	Workgroup forwarding all calls.
3	ForwardBusy	All logged in agents on call.
4	ForwardNoAnswer	All available agents did not answer.
5	FwdNoLoginAgent	No logged in agents.

Table B-28 Queue Call Exit Reason Enumeration Table

ExitReason	Name	Description
6	Reserved	
7	Abandon	Call dropped while in WG or Queue.
8	Reserved	
9	TransferVM	Option taken to transfer to voice mail
10	TransferExtension	Option taken to transfer to an extension.
11	TransferMenu	Option taken to transfer to a menu.
12	Pickup	Agent picked up call from queue.
13	Unpark	Agent parked call from queue.

- ForwardMaxRings is no longer used.
- Exit Reasons for Forwarding (2-5) reflects the call being forwarded from the workgroup. These are used when the call leaves the workgroup as a result of call handling and the call handling indicates to forward the call to an internal or external number. Call handling can also indicate that the call is entering the call queue for the workgroup. In that case, these exit reasons are not used because the call does not exit the queue at that point.
- Exit Reason 8, Abandon, is used when the caller drops the call either by physically hanging up or by taking an option on a Queue Step to hang up.
- Even after a call is forwarded to the queue, it remains on the queue and it may still be successfully transferred to an agent or abandoned. Exit Reason 1 or 7 is recorded if either of these occurs.
- In addition to a call being successfully hunted or abandoned while on the queue, it may exit the queue because of an option taken during a queue step. The call will exit the queue if the caller takes any of the following options:
  — Take a message
  — Transfer to extension
  — Go to menu
  — Exit reasons 9, 10, and 11 have been added to cover these cases.

Table B-29 describes the elements in the Queue Call Target Type Enumeration table.

Table B-29 Queue Call Target Type Enumeration Table

Target #	TargetType	Description
0	Null	
1	Agent	Workgroup agent.
2	Menu	A menu on the ShoreTel system.
3	Mailbox	A mailbox on the ShoreTel system.
4	OtherIntrnExtrn	Any other extensions to which the call is targeted.

B.11 QueueStep Table

The QueueStep table logs data about time spent in queue steps or in hunting for agents. Table B-30 describes the elements in the QueueStep table.

Table B-30 Queue Call Field Descriptions

Field Name	Data Type	Description
ID	AutoNumber	Unique identifier. (4-byte integer, required)
QcallTableID	Number	Link to the Autonumber field in the QueueCall table. This essentially identifies the QueueCall that this step is associated with. (4-byte integer)
StartTime	Date/Time	Time at which the call first enters this step. (8-byte date/time)
Duration	Date/Time	Elapsed time spent in this step. (8-byte date/time)
DurationSeconds	Number	Elapsed seconds spent in this step. (4-byte integer)
StepNumber	Number	Step number if this is not a hunting record (as identified by the Hunting field set to Yes). The step number corresponds to the step number in the workgroup configuration.
ExitReason	Number	Enumerated reason for exit from step. (1-byte integer)
Hunting	Yes/No	If true the times correspond to hunting, or else this indicates a queue step. (boolean)

There is a record for each period that the call spends hunting and for each period a call spends in a queue step. For example, if a call to a workgroup initially hunts for agents, then goes to the queue and exits the workgroup from that queue step, there will be two records for the call in the QueueStep table. The first record would be for hunting (the duration may be zero if, for example, no agents were logged in). The second record is for the first queue step from which the call exited.

B.12 Web Tables

Web tables log call data for audio only, web only, and audio and web conferences.

Audio Only Conference:

- The Connect table includes a record for each leg of the conference.
- All legs in the same conference have a similar Call.CallID.
- No records are added to web_session and web_attendee tables

Web Only Conference:

- A meeting session record is written to the web_session table after the meeting ends.
- A record is written to the web_attendee table for each attendee.
- No records are added to the Call and Connect tables.

Audio & Web Conference:

- The Connect table includes a record for each leg of the conference.
- All legs in the same conference have a similar Call.CallID.
- A meeting session record is written to the web_session table after the meeting ends.

- A record is written to the web_attendee table for each attendee.
- The Web_session table holds a CallID that references Call.CallID. If a web attendee reconciles his or her audio leg, web_attendee.caller_id references Connect.PartyID.

B.12.1 Web Session Table

Table B-31 provides information about the elements that appear in the log for web sessions.

Table B-31 Elements in the Web Session Log

Field Name	Data Type	Description
Id	AutoNumber	The session ID referenced by the attendee table
meeting_title	Text	Title of the meeting
meeting_desc	Text	Description of the meeting
meeting_type	Text	Normal, open or panel
start_time	Date/Time	Local time of UCB server when meeting session started
start_local_time	Date/Time	Local time of HQ server when meeting session started
start_utc_time	Date/Time	UTC time meeting session started
start_local_dst_flag	Yes/No	Flag to set UCB server dst flag
start_hq_dst_flag	Yes/No	Flag to set HQ dst flag
end_time	Date/Time	Local time of UCB server when meeting session ended
end_local_time	Date/Time	Local time of HQ server when meeting session ended
end_utc_time	Date/Time	UTC time meeting session ended
end_local_dst_flag	Yes/No	Flag to set local dst flag of UCB server
end_hq_dst_flag	Yes/No	Flag to set HQ dst flag
host_login	Text	Login name of meeting host
login_type	Text	Name, password, registration, none
Scheduled	Yes/No	'Y' or 'N'
Public	Yes/No	'Y' or 'N'
Server	Text	IP address of conference bridge
moderator_code	Text	Moderator's access code
Attendee_code	Text	Attendee's access code
CallID	Number	CallID associated with conference call. NULL for web only conference.
Archived	Yes/No	Has this entry been archived?

B.12.2 Web Attendee Table

Table B-32 provides information about the elements that appear in the log for web session attendees.

Table B-32 Elements In Web Attendee Log

Field Name	Data Type	Description
session_id	Number	References ID from web_session table
start_time	Date/Time	Local time of UCB server when attendee joined session
start_local_time	Date/Time	Local time of HQ server when attendee joined session
start_utc_time	Date/Time	UTC time attendee joined session
start_local_dst_flag	Yes/No	Flag to set UCB server dst flag
start_hq_dst_flag	Yes/No	Flag to set HQ dst flag
end_time	Date/Time	Local time of UCB server when attendee left session
end_local_time	Date/Time	Local time of HQ server when attendee left session
end_utc_time	Date/Time	UTC time attendee left session
end_local_dst_flag	Yes/No	Flag to set UCB server dst flag
end_hq_dst_flag	Yes/No	Flag to set HQ dst flag
break_time	Number	Time attendee is absent from meeting
user_name	Text	Login name of attendee
user_ip	Text	Attendee's IP address
caller_id	Text	Caller's phone number

Appendix C

ShoreTel Switches

This appendix describes the ShoreTel Voice Switches. The topics discussed include:

- "Switch Models" on page 675
- "Specifications for ShoreTel 1U Half-Width Voice Switches" on page 678
- "Specifications for ShoreTel Voice Model Switches" on page 714
- "Specification for ShoreTel 1U Full-Width Voice Switches" on page 725

C.1 Switch Models

Switch model numbers are located on the rear panel as shown in Figure C-1.

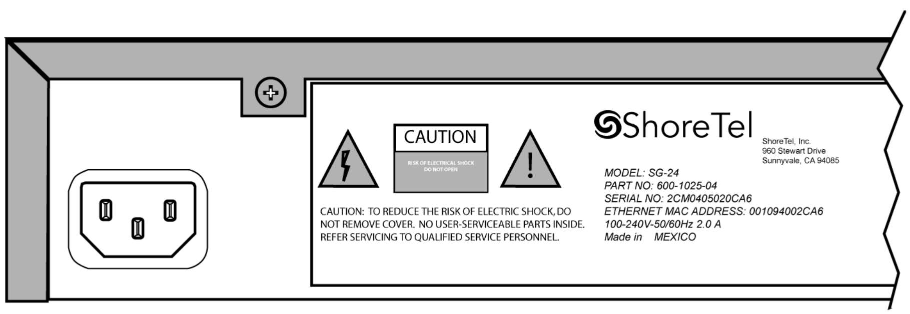


Figure C-1 Switch Model Number Label

ShoreTel models are classified into two switch families, based on chassis type:

- 1-U Half Width Switches
- 1-U Full Width Switches

The following is a brief description of each switch family.

C.1.1 ShoreTel 1-U Half Width Voice Switches

ShoreTel Voice Switches are classified into two switch families, based on chassis type:

- 1-U Half-width switches
- 1-U Full-width switches

The next section briefly describes each switch family.

C.1.2 ShoreTel 1-U Half-Width Voice Switches

The ShoreTel 1-U Half-Width Switch family is the most recent ShoreTel Voice Switch design. The 1-U Half Width has a smaller footprint, uses less power, and has lower heat dissipation requirements than earlier ShoreTel Voice Switches. These switches offer higher granularity in the number of IP users supported, allowing customers to program the switch to satisfy their precise requirements.

The switches can be stacked or mounted in a standard 19-inch rack. Rack mounting 1-U Half Width Switches requires the ShoreTel Dual Tray. One or two switches are inserted into the Dual Tray, which is then mounted into the 19-inch rack. Two switches are mounted side by side. Rack mounting the switches require the ShoreTel Dual Tray.

ShoreTel 1-U Half Width Voice Switch models include:

- ShoreTel Voice Switch 30
- ShoreTel Voice Switch 30BRI
- ShoreTel Voice Switch 50
- ShoreTel Voice Switch 90
- ShoreTel Voice Switch 90BRI
- ShoreTel Voice Switch 220T1
- ShoreTel Voice Switch 220T1A
- ShoreTel Voice Switch T1k
- ShoreTel Voice Switch 220E1
- ShoreTel Voice Switch E1k

C.1.3 ShoreTel Voicemail Model Voice Switches

Voicemail Model Switches are ShoreTel Voice Switch that provide voicemail services and access to auto attendant menus for extensions hosted by the switch. Voicemail Model (V Model) switches provide local access to voicemail while being controlled by a Distributed server at a different location.

The switches can be stacked or mounted in a standard 19-inch rack. Rack mounting 1-U Half Width Switches requires the ShoreTel Dual Tray. One or two switches are inserted into the Dual Tray, which is then mounted into the 19-inch rack. Two switches are mounted side by side Rack mounting the switches require the ShoreTel Dual Tray

ShoreTel V Model Switch models include:

- ShoreTel Voice Switch 90V
- ShoreTel Voice Switch 50V
- ShoreTel Voice Switch 90BRIV

C.1.3.1 Capacity

Number of V Model switches allowed per system

A ShoreTel system supports a maximum of 100 V Model Switches. There are no restrictions concerning the allocation of switches among the sites defined by the system.

Simultaneous Voicemail Calls per V Model switches

Voicemail Model Switches support the following number of simultaneous voicemail calls.

- ShoreTel Voice Switch 50V – Maximum of 5 Voicemail calls per switch

— G.711 calls: 5
— G.729 calls: 2

- ShoreTel Voice Switch 90V – Maximum of 9 Voicemail calls per switch
  — G7.11 calls: 9
  — G7.29 calls: 4
- ShoreTel Voice Switch 90BRIV – Maximum of 9 Voicemail calls per switch
  — G.711 calls: 9
  — G.729 calls: 4

Call Load

Voicemail Model Switches call load capacity is as follows:

- 5400 BHCC when supporting 90 MGCP IP Phones or 90 SIP Trunks
- 3600 BHCC when supporting 90 SIP IP Phones or 90 SIP Trunks

Compact Flash Memory

Voicemail Model switches store voicemail and Auto Attendant files on compact flash. Flash card capacity for V Model Switches is:

- ShoreTel Voice Switch 50V: 1 GB
- ShoreTel Voice Switch 90V: 2 GB
- ShoreTel Voice Switch90 BRIV: 2 GB

Media Support

Voicemail Model Switches support the following media streams:

- G.711
  — Music on Hold (MOH): 15 calls
  — Backup Auto Attendant (BAA): 50 calls
- G.729
  — Music on Hold (MOH): none
  — Backup Auto Attendant (BAA): none

SIP support

Voicemail Model Switches support the following SIP media streams:

- G.711 Ringback tone (Hunt Groups and Work Group calls): 50 media streams
- G.729 Ringback tone (Hunt Groups and Work Group calls): no support

C.1.4 ShoreTel 1-U Full Width Voice Switches

The ShoreTel 1-U Full Width Switch family includes three models that support analog, IP, SIP, T1, and E1 voice data streams. Full width switch models can be stacked or mounted directly into a standard 19-inch equipment rack. These switches are all 1 RU and have an RJ21X connector for connection to analog phones and trunks. They also feature redundant Ethernet LAN connections for greater availability and reliability.

ShoreTel 1-U Full Width Voice Switch models include:

- ShoreTel Voice Switch 120 – also referred to as ShoreTel Voice Switch 120/24
- ShoreTel Voice Switch 60 – also referred to as ShoreTel Voice Switch 60/12
- ShoreTel Voice Switch 40 – also referred to as ShoreTel Voice Switch 40/8
- ShoreTel Voice Switch T1
- ShoreTel Voice Switch E1
- ShoreTel Voice Switch 24A

C.2 Specifications for ShoreTel 1U Half-Width Voice Switches

C.2.1 ShoreTel Voice Switch 90

The following sections describe ShoreTel Voice Switch 90 resource capacity, LED behavior, and connectors. The ShoreTel Voice Switch 90 is not supported in installations outside the U.S. and Canada. Figure C-2 displays the ShoreTel Voice Switch 90 front plate.

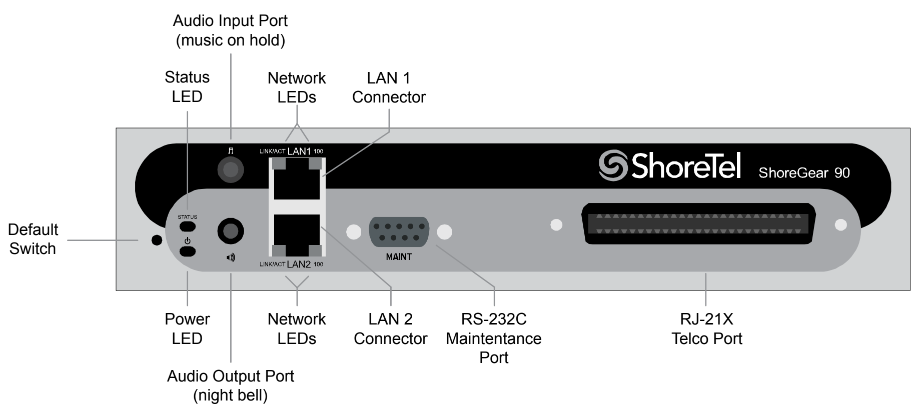


Figure C-2 ShoreTel Voice Switch 90 Front Plate

C.2.1.1 Switch Capacity

- **Analog Circuit Resources**
  - **Ports 1-8:** Eight Loop Start Trunks
  - **Ports 9-12:** Four Extensions or DID Trunks. A single command configures all ports as either Extensions or DID trunks.
  - **Power Failure Transfer Unit:** Trunk Port 1 to Extension Port 12
- **Make Me Conference Resources:** 12 ports
  - **Ports 1-12**
- **Maximum IP Phone Resources:** 90 devices
  - **Analog Port Reallocation:** 60
  - **Built-in Resources:** 30

C.2.1.2 LED Descriptions

Power LED

The ShoreTel Voice Switch 90 has one power LED It indicates the following:

- **On**: The switch is operating normally.
- **Off**: The switch has no power.
- **Flashing**:
  - — **2 flashes—**The switch failed its internal self-test. This indicates a hardware failure. Replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
  - — **3 flashes—**Booting via FTP. Flash memory might be corrupted. Go to the Quick Look page to ensure that the system is running properly.
  - — **4 flashes—**The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.
  - — **5 flashes—**The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreTel server, and set the boot file to `/tsk/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreTel server's IP address, and set option 67 to `/tsk/vxworks`.
  - — **6 flashes—**Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

The ShoreTel Voice Switch 90 network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:
  - — **When flashing**, network activity is detected.
  - — **When on (not flashing)**, the switch is connected to an Ethernet network.
  - — **When off**, the switch cannot detect an Ethernet network.

This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- **100M:**
  - — **When green**, the switch is connected to a 100BaseT network.
  - — **When off**, the switch is connected to a 10BaseT network.

Status LED

The ShoreTel Voice Switch 90 has one status LED to provide general information about the ports. The color and blink pattern of the LED indicate the port function:

- **Status LED (Green)**
  - — **When on steady**, no ports are handling active calls.
  - — **When flashing fast**, at least one port is handling an active call.
- **Status LED (Yellow)**
  - — **When on steady**, no ports are handling active calls and at least one port is out of service.
  - — **When flashing slow**, the switch is not connected (or has lost connection) to a ShoreTel server.
  - — **When flashing fast**, at least one port is handling an active call and at least one port is out of service.
- **Off:** No ports are assigned.

C.2.1.3 ShoreTel Voice Switch 90 Connectors

The ShoreTel Voice Switch 90 voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-21X male connector for mass termination of the telephone/trunk ports
  - — **Power Failure Transfer Unit:** Trunk Port 1 to Extension Port 12
  - — **Backup Operator:** Extension Port 12

ShoreTel Voice Switch 90 RJ-21X Telephone and Trunk Connector

Table C-1 lists the RJ-21X Ring and Tip pin numbers for the SG 90.

Table C-1 ShoreTel Voice Switch 90 RJ-21X Telephone and Trunk Connector Pins

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
1	Trunk	1	Blue/White	26	White/Blue
–		2	Orange/White	27	White/Orange
2	Trunk	3	Green/White	28	White/Green
–		4	Brown/White	29	White/Brown
3	Trunk	5	Slate/White	30	White/Slate
–		6	Blue/Red	31	Red/Blue
4	Trunk	7	Orange/Red	32	Red/Orange
–		8	Green/Red	33	Red/Green
5	Trunk	9	Brown/Red	34	Red/Brown
–		10	Slate/Red	35	Red/Slate
6	Trunk	11	Blue/Black	36	Black/Blue
–		12	Orange/Black	37	Black/Orange
7	Trunk	13	Green/Black	38	Black/Green
–		14	Brown/Black	39	Black/Brown
8	Trunk	15	Slate/Black	40	Black/Slate
–		16	Blue/Yellow	41	Yellow/Blue
9	Extension - DID	17	Orange/Yellow	42	Yellow/Orange
–		18	Green/Yellow	43	Yellow/Green
10	Extension - DID	19	Brown/Yellow	44	Yellow/Brown
–		20	Slate/Yellow	45	Yellow/Slate
11	Extension - DID	21	Blue/Violet	46	Violet/Blue
–		22	Orange/Violet	47	Violet/Orange
12	Extension - DID	23	Green/Violet	48	Violet/Green
–		24	Brown/Violet	49	Violet/Brown
–		25	Slate/Violet	50	Violet/Slate

C.2.2 ShoreTel Voice Switch 90BRI

The following sections describe ShoreTel Voice Switch 90BRI resource capacity, LED behavior, and connectors. Figure C-3 displays the ShoreTel Voice Switch 90BRI front plate.

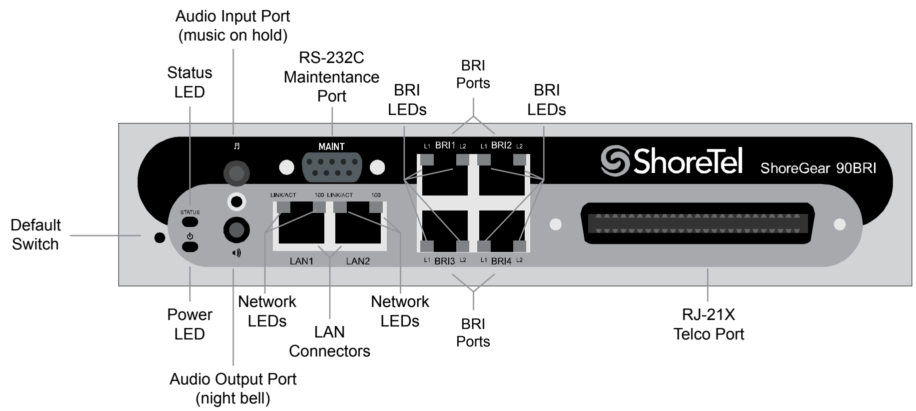


Figure C-3 ShoreTel Voice Switch 90BRI Front Plate

C.2.2.1 Switch Capacity

- **Analog Circuit Resources**
  - — **Ports 9-12**: Extensions
- **Digital Circuit Resources**
  - — **Four BRI Spans, each comprising two channels**: Eight channels maximum
- **Make Me Conference Resources:** 4 ports
  - — **Ports 9-12**
- **Maximum IP Phone Resources**: 90 devices
  - — **Analog Port Reallocation:** 20
  - — **Digital Channel Reallocation**: 40
  - — **Built-in Resources**: 30

C.2.2.2 LED Descriptions

Power LED

The ShoreTel Voice Switch 90BRI has one power LED, which indicates the following:

- **On**: The switch is operating normally.
- **Off**: The switch has no power.
- **Flashing**
  - — **2 flashes—**The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
  - — **3 flashes—**Booting via FTP. Flash memory might be corrupted. Go to the Quick Look page to ensure that the system is running properly.
  - — **4 flashes—**The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds

and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.

— **5 flashes—**The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreTel server, and set the boot file to `/tsk/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreTel server's IP address, and set option 67 to `/tsk/vxworks`.

— **6 flashes—**Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

The ShoreTel Voice Switch 90BRI network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:
  — **When flashing**, network activity is detected.
  — **When on (not flashing)**, the switch is connected to an Ethernet network.
  — When off, the switch cannot detect an Ethernet network.

  This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.
- **100M**
  — **When green**, the switch is connected to a 100BaseT network.
  — **When off,** the switch is connected to a 10BaseT network.

Status LED

The ShoreTel Voice Switch 90BRI has one status LED to provide general information about the ports. The color and blink pattern of the LED indicate the port function:

- **Status LED (Green)**
  — **When on steady**, no ports are handling active calls.
  — **When flashing fast (100 msec on/off)**, at least one port is handling an active call.
- **Status LED (Yellow)**

— When on steady, no ports are handling active calls and at least one port is out of service.

— **When flashing slow (1 sec. on/off)**, the switch is not connected (or has lost connection) to a ShoreTel server.

— **When flashing fast (100 msec on/off)**, at least one port is handling an active call and at least one port is out of service.

- **Off:** No ports are assigned.

BRI LED

Each BRI connector has two LEDs to indicate port activity. The color and blink pattern of the LED indicate the port function:

- **LED 1: Off, LED 2 Off –** Port not configured in Director
- **LED 1: Yellow, LED 2 Off –** Port inactive or not connected
- **LED 1: Off, LED 2 Off –** Layer 1 active. Layer 2 not established
- **LED 1: Off, LED 2 Green –** Layer 1 active. Layer 2 active.
- **LED 1: Off, LED 2 Green flashing –** Call in progress (Layer 1, Layer 2, and Layer 3 active).

C.2.2.3 ShoreTel Voice Switch 90BRI Connectors

The ShoreTel Voice Switch 90BRI voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-21X male connector for mass termination of the telephone/trunk ports
- 4 RJ-45 T1 telco port

ShoreTel Voice Switch 90BRI RJ-21X Telephone and Trunk Connector

Table C-2 lists the RJ-21X Ring and Tip pin numbers for the SG 90BRI

Table C-2 ShoreTel Voice Switch 90BRI RJ-21X Telephone and Trunk Connector Pins

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
–		1	Blue/White	26	White/Blue
–		2	Orange/White	27	White/Orange
–		3	Green/White	28	White/Green
–		4	Brown/White	29	White/Brown
–		5	Slate/White	30	White/Slate
–		6	Blue/Red	31	Red/Blue

Table C-2 ShoreTel Voice Switch 90BRI RJ-21X Telephone and Trunk Connector Pins (Continued)

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
–		7	Orange/Red	32	Red/Orange
–		8	Green/Red	33	Red/Green
–		9	Brown/Red	34	Red/Brown
–		10	Slate/Red	35	Red/Slate
–		11	Blue/Black	36	Black/Blue
–		12	Orange/Black	37	Black/Orange
–		13	Green/Black	38	Black/Green
–		14	Brown/Black	39	Black/Brown
–		15	Slate/Black	40	Black/Slate
–		16	Blue/Yellow	41	Yellow/Blue
9	Extension	17	Orange/Yellow	42	Yellow/Orange
–		18	Green/Yellow	43	Yellow/Green
10	Extension	19	Brown/Yellow	44	Yellow/Brown
–		20	Slate/Yellow	45	Yellow/Slate
11	Extension	21	Blue/Violet	46	Violet/Blue
–		22	Orange/Violet	47	Violet/Orange
12	Extension	23	Green/Violet	48	Violet/Green
–		24	Brown/Violet	49	Violet/Brown
–		25	Slate/Violet	50	Violet/Slate

C.2.3 ShoreTel Voice Switch 50

The following sections describe ShoreTel Voice Switch 50 resource capacity, LED behavior, and connectors. The ShoreTel Voice Switch 50 is not supported in installations outside the U.S. and Canada. Figure C-4 displays the ShoreTel Voice Switch 50 faceplate.

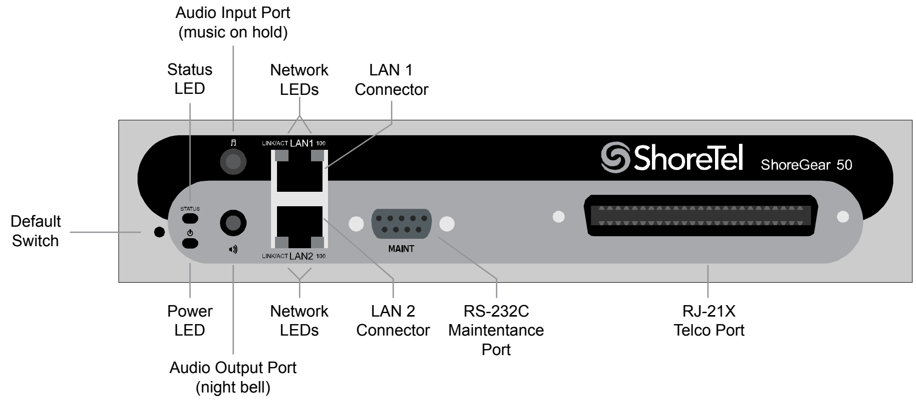


Figure C-4 ShoreTel Voice Switch 50 Front Plate

C.2.3.1 Switch Capacity

- **Analog Circuit Resources**
  - **Ports 1-4**: Four Loop Start Trunks
  - **Ports 11-12:** Two Extensions or DID Trunks. A single command configures all ports as either Extensions or DID trunks.
  - **Power Failure Transfer Unit:** Trunk Port 1 to Extension Port 12
- **Make Me Conference Resources:** six ports
  - **Ports 1-4, 11-12**
- **Maximum IP Phone Resources:** 50 devices
  - **Analog Port Reallocation:** 30
  - **Built-in Resources:** 20

C.2.3.2 LED Descriptions

Power LED

The ShoreTel Voice Switch 50 has one power LED, which indicates the following:

- **On**: The switch is operating normally.
- **Off**: The switch has no power.
- **Flashing**
  - **2 flashes—**The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
  - **3 flashes—**Booting via FTP. Flash memory might be corrupted. Go to the Quick Look page to ensure that the system is running properly.
  - **4 flashes—**The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.

— **5 flashes—**The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreTel server, and set the boot file to `/tsk/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreTel server's IP address, and set option 67 to `/tsk/vxworks`.

— **6 flashes—**Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

The ShoreTel Voice Switch 50 network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:

  — **When flashing**, network activity is detected.

  — **When on (not flashing)**, the switch is connected to an Ethernet network.

  — **When off**, the switch cannot detect an Ethernet network.

  This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- **100M**

  — **When green**, the switch is connected to a 100BaseT network.

  — **When off**, the switch is connected to a 10BaseT network.

Status LED

The ShoreTel Voice Switch 50 has one status LED to provide general information about the ports. The color and blink pattern of the LED indicate the port function:

- **Status LED (Green)**

  — **When on steady**, no ports are handling active calls.

  — **When flashing fast**, at least one port is handling an active call.

- **Status LED (Yellow)**

  — **When on steady**, no ports are handling active calls and at least one port is out of service.

— **When flashing slow**, the switch is not connected (or has lost connection) to a ShoreTel server.

— **When flashing fast**, at least one port is handling an active call and at least one port is out of service.

- **Off:** No ports are assigned.

C.2.3.3 ShoreTel Voice Switch 50 Connectors

The ShoreTel Voice Switch 50 voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-21X male connector for mass termination of the telephone/trunk ports
  - — **Power Failure Transfer Unit:** Trunk Port 1 to Extension Port 12
  - — **Backup Operator:** Extension Port 12

ShoreTel Voice Switch 50 RJ-21X Telephone and Trunk Connector

Table C-3 lists the RJ-21X Ring and Tip pin numbers for the SG 50.

Table C-3 ShoreTel Voice Switch 50 RJ-21X Telephone and Trunk Connector Pins

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
1	Trunk	1	Blue/White	26	White/Blue
–		2	Orange/White	27	White/Orange
2	Trunk	3	Green/White	28	White/Green
–		4	Brown/White	29	White/Brown
3	Trunk	5	Slate/White	30	White/Slate
–		6	Blue/Red	31	Red/Blue
4	Trunk	7	Orange/Red	32	Red/Orange
–		8	Green/Red	33	Red/Green
–		9	Brown/Red	34	Red/Brown
–		10	Slate/Red	35	Red/Slate
–		11	Blue/Black	36	Black/Blue
–		12	Orange/Black	37	Black/Orange
–		13	Green/Black	38	Black/Green
–		14	Brown/Black	39	Black/Brown
–		15	Slate/Black	40	Black/Slate
–		16	Blue/Yellow	41	Yellow/Blue

Table C-3 ShoreTel Voice Switch 50 RJ-21X Telephone and Trunk Connector Pins (Continued)

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
9	Extension - DID	17	Orange/Yellow	42	Yellow/Orange
–		18	Green/Yellow	43	Yellow/Green
10	Extension - DID	19	Brown/Yellow	44	Yellow/Brown
–		20	Slate/Yellow	45	Yellow/Slate
11	Extension - DID	21	Blue/Violet	46	Violet/Blue
–		22	Orange/Violet	47	Violet/Orange
12	Extension - DID	23	Green/Violet	48	Violet/Green
–		24	Brown/Violet	49	Violet/Brown
–		25	Slate/Violet	50	Violet/Slate

C.2.4 ShoreTel Voice Switch 30

The following sections describe ShoreTel Voice Switch 30 resource capacity, LED behavior, and connectors. The ShoreTel Voice Switch 30 is not supported in installations outside the U.S. and Canada. Figure C-5 displays the ShoreTel Voice Switch 30 front plate.

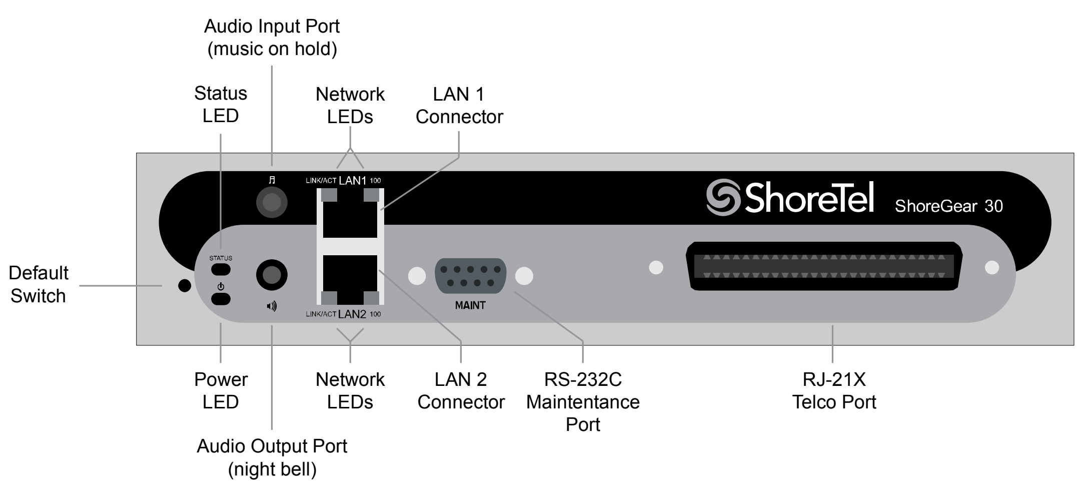


Figure C-5 ShoreTel Voice Switch 30 Front Plate

C.2.4.1 Switch Capacity

- Analog Circuit Resources
  — **Ports 1-2:** Two Loop Start Trunks

   — **Ports 11-12:** Two Extensions or DID Trunks. A single command configures all ports as either Extensions or DID trunks.
   — **Power Failure Transfer Unit:** Trunk Port 1 to Extension Port 12
- **Make Me Conference Resources:** none
- **Maximum IP Phone Resources:** none
   — **Analog Port Reallocation:** 20
   — **Built-in Resources:** 10

C.2.4.2 LED Descriptions

Power LED

The ShoreTel Voice Switch 30 has one power LED, which indicates the following:

- **On:** The switch is operating normally.
- **Off:** The switch has no power.
- **Flashing:**
   — **2 flashes—**The switch failed its internal self-test. This indicates a hardware failure. Replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
   — **3 flashes—**Booting via FTP. Flash memory might be corrupted. Go to the Quick Look page to ensure that the system is running properly.
   — **4 flashes—**The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.
   — **5 flashes—**The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreTel server, and set the boot file to `/tsk/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreTel server's IP address, and set option 67 to `/tsk/vxworks`.
   — **6 flashes—**Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

The ShoreTel Voice Switch 30 network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:
  - **When flashing**, network activity is detected.
  - **When on (not flashing)**, the switch is connected to an Ethernet network.
  - **When off**, the switch cannot detect an Ethernet network.

  This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.
- **100M:**
  - **When green**, the switch is connected to a 100BaseT network.
  - **When off**, the switch is connected to a 10BaseT network.

Status LED

The ShoreTel Voice Switch 30 has one status LED to provide general information about the ports. The color and blink pattern of the LED indicate the port function:

- **Status LED (Green)**
  - **When on steady**, no ports are handling active calls.
  - **When flashing fast**, at least one port is handling an active call.
- **Status LED (Yellow)**
  - **When on steady**, no ports are handling active calls and at least one port is out of service.
  - **When flashing slow**, the switch is not connected (or has lost connection) to a ShoreTel server.
  - **When flashing fast**, at least one port is handling an active call and at least one port is out of service.
- **Off:** No ports are assigned.

C.2.4.3 ShoreTel Voice Switch 30 Connectors

The ShoreTel Voice Switch 30 voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-21X male connector for mass termination of the telephone/trunk ports
  - **Power Failure Transfer Unit:** Trunk Port 1 to Extension Port 12
  - **Backup Operator:** Extension Port 12

ShoreTel Voice Switch 30 RJ-21X Telephone and Trunk Connector

Table C-4 lists the RJ-21X Ring and Tip pin numbers for the SG 30.

Table C-4 ShoreTel Voice Switch 30 RJ-21X Telephone and Trunk Connector Pins

		Ring		Tip	
Port	Type	Pin #	Cable Color	Pin #	Cable Color
1	Trunk	1	Blue/White	26	White/Blue
–		2	Orange/White	27	White/Orange
2	Trunk	3	Green/White	28	White/Green
–		4	Brown/White	29	White/Brown
–		5	Slate/White	30	White/Slate
–		6	Blue/Red	31	Red/Blue
–		7	Orange/Red	32	Red/Orange
–		8	Green/Red	33	Red/Green
–		9	Brown/Red	34	Red/Brown
–		10	Slate/Red	35	Red/Slate
–		11	Blue/Black	36	Black/Blue
–		12	Orange/Black	37	Black/Orange
–		13	Green/Black	38	Black/Green
–		14	Brown/Black	39	Black/Brown
–		15	Slate/Black	40	Black/Slate
–		16	Blue/Yellow	41	Yellow/Blue
–		17	Orange/Yellow	42	Yellow/Orange
–		18	Green/Yellow	43	Yellow/Green
–		19	Brown/Yellow	44	Yellow/Brown
–		20	Slate/Yellow	45	Yellow/Slate
11	Extension - DID	21	Blue/Violet	46	Violet/Blue
–		22	Orange/Violet	47	Violet/Orange
12	Extension - DID	23	Green/Violet	48	Violet/Green
–		24	Brown/Violet	49	Violet/Brown
–		25	Slate/Violet	50	Violet/Slate

C.2.5 ShoreTel Voice Switch 30BRI

The following sections describe ShoreTel Voice Switch 30BRI resource capacity, LED behavior, and connectors. Figure C-6 displays the ShoreTel Voice Switch 30BRI front plate.

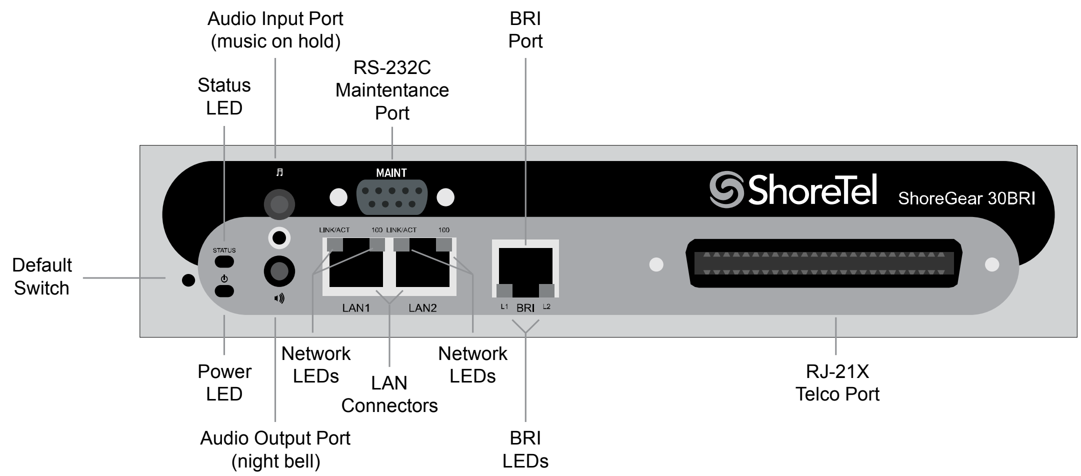


Figure C-6 ShoreTel Voice Switch 30BRI Front Plate

C.2.5.1 Switch Capacity

- **Analog Circuit Resources**
  - — **Ports 11-12:** Extensions
- **Digital Circuit Resources**
  - — **One BRI Span comprising two channels:** two channels maximum
- **Make Me Conference Resource:** None
- **Maximum IP Phone Resources:** 30 devices
  - — **Analog Port Reallocation:** 10
  - — **Digital Channel Reallocation:** 10
  - — **Built-in Resources:** 10

C.2.5.2 LED Descriptions

Power LED

The ShoreTel Voice Switch 30BRI has one power LED, which indicates the following:

- **On:** The switch is operating normally.
- **Off:** The switch has no power.
- **Flashing**
  - — **2 flashes—**The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
  - — **3 flashes—**Booting via FTP. Flash memory might be corrupted. Go to the Quick Look page to ensure that the system is running properly.
  - — **4 flashes—**The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory

to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.

— **5 flashes—**The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreTel server, and set the boot file to `/tsk/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreTel server's IP address, and set option 67 to `/tsk/vxworks`.

— **6 flashes—**Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

The ShoreTel Voice Switch 30BRI network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:

  — When flashing, network activity is detected.

  — When on (not flashing), the switch is connected to an Ethernet network.

  — When off, the switch cannot detect an Ethernet network.

  This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- 100M

  — **When green**, the switch is connected to a 100BaseT network.

  — **When off**, the switch is connected to a 10BaseT network.

Status LED

The ShoreTel Voice Switch 30BRI has one status LED to provide general information about the ports. The color and blink pattern of the LED indicate the port function:

- Status LED (Green)

  — **When on steady**, no ports are handling active calls.

  — **When flashing fast (100 msec on/off)**, at least one port is handling an active call.

- **Status LED (Yellow)**
  - — **When on steady**, no ports have active calls, and at least one port is out of service.
  - — **When flashing slow (1 sec. on/off)**, the switch is not connected (or has lost connection) to a ShoreTel server.
  - — **When flashing fast (100 msec on/off)**, at least one port is handling an active call and at least one port is out of service.
- **Off**: No ports are assigned.

BRI LED

Each BRI connector has two LEDs to indicate port activity. The color and blink pattern of the LED indicate the port function:

- **LED 1: Off, LED 2 Off –** Port not configured in Director
- **LED 1: Yellow, LED 2 Off –** Port inactive or not connected
- **LED 1: Off, LED 2 Off –** Layer 1 active. Layer 2 not established
- **LED 1: Off, LED 2 Green –** Layer 1 active. Layer 2 active.
- **LED 1: Off, LED 2 Green flashing –** Call in progress (Layer 1, Layer 2, and Layer 3 active).

C.2.5.3 ShoreTel Voice Switch 30BRI Connectors

The ShoreTel Voice Switch 30BRI voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-21X male connector for mass termination of the telephone/trunk ports
- 4 RJ-45 T1 telco port

ShoreTel Voice Switch 30BRI RJ-21X Telephone and Trunk Connector

Table C-5 lists the RJ-21X Ring and Tip pin numbers for the SG 30BRI.

Table C-5 ShoreTel Voice Switch 30BRI RJ-21X Telephone and Trunk Connector Pins

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
–		1	Blue/White	26	White/Blue
–		2	Orange/White	27	White/Orange
–		3	Green/White	28	White/Green
–		4	Brown/White	29	White/Brown
–		5	Slate/White	30	White/Slate

Table C-5 ShoreTel Voice Switch 30BRI RJ-21X Telephone and Trunk Connector Pins (Continued)

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
–		6	Blue/Red	31	Red/Blue
–		7	Orange/Red	32	Red/Orange
–		8	Green/Red	33	Red/Green
–		9	Brown/Red	34	Red/Brown
–		10	Slate/Red	35	Red/Slate
–		11	Blue/Black	36	Black/Blue
–		12	Orange/Black	37	Black/Orange
–		13	Green/Black	38	Black/Green
–		14	Brown/Black	39	Black/Brown
–		15	Slate/Black	40	Black/Slate
–		16	Blue/Yellow	41	Yellow/Blue
9	Extension	17	Orange/Yellow	42	Yellow/Orange
–		18	Green/Yellow	43	Yellow/Green
10	Extension	19	Brown/Yellow	44	Yellow/Brown
–		20	Slate/Yellow	45	Yellow/Slate
11	Extension	21	Blue/Violet	46	Violet/Blue
–		22	Orange/Violet	47	Violet/Orange
12	Extension	23	Green/Violet	48	Violet/Green
–		24	Brown/Violet	49	Violet/Brown
–		25	Slate/Violet	50	Violet/Slate

C.2.6 ShoreTel Voice Switch 220T1

The following sections describe ShoreTel Voice Switch 220T1 resource capacity, LED behavior, and connectors. The ShoreTel Voice Switch 220T1 is not supported in installations outside the U.S. and Canada. Figure C-7 displays the ShoreTel Voice Switch 220T1 front plate.

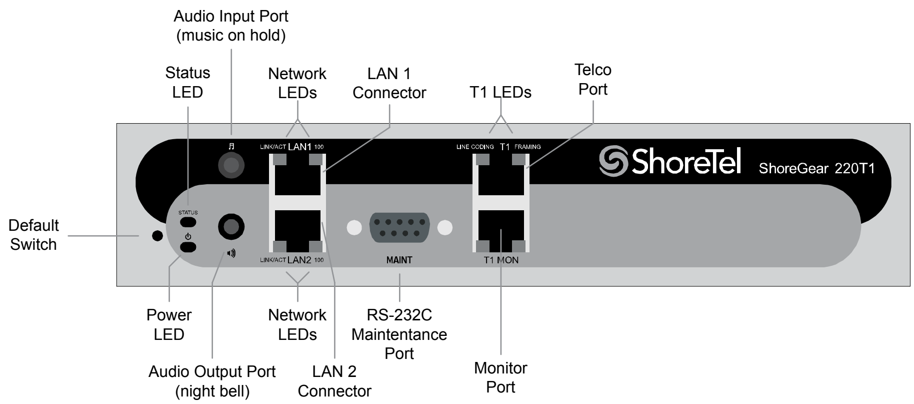


Figure C-7 ShoreTel Voice Switch 220T1 Front Plate

C.2.6.1 Switch Capacity

- **Digital Circuit Resources**: 24 channels maximum
  - **One T1 circuit, 24 channels per circuit**: 24 channels maximum
- **Make Me Conference Resource:** None
- **Maximum IP Phone Resources**: 220
  - **Digital Channel Reallocation**: 120
  - **Built-in Resources**: 100

C.2.6.2 LED Descriptions

Power LED

The ShoreTel Voice Switch 220T1 has one power LED, which indicates the following:

- **On**: The switch is operating normally.
- **Off**: The switch has no power.
- **Flashing**
  - **2 flashes—**The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
  - **3 flashes—**Booting via FTP. Flash memory might be corrupted. Go to the Quick Look page to ensure that the system is running properly.
  - **4 flashes—**The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.
  - **5 flashes—**The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreTel server, and set

the boot file to `/tsk/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreTel server's IP address, and set option 67 to `/tsk/vxworks`.

— **6 flashes—**Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

The ShoreTel Voice Switch 220T1 network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:
  - **When flashing**, network activity is detected.
  - **When on (not flashing)**, the switch is connected to an Ethernet network.
  - **When off**, the switch cannot detect an Ethernet network.

  This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.
- **100M**
  - **When green**, the switch is connected to a 100BaseT network.
  - **When off**, the switch is connected to a 10BaseT network.

Status LED

TheShoreTel Voice Switch 220T1 has one status LED to provide general information about the ports. The color and blink pattern of the LED indicate the port function:

- **Status LED (Green)**
  - **When on steady**, no ports are handling active calls.
  - **When flashing fast**, at least one port is handling an active call.
- **Status LED (Yellow)**
  - **When on steady**, no ports are handling active calls and at least one port is out of service.
  - **When flashing slow**, the switch is not connected (or has lost connection) to a ShoreTel server.

— **When flashing fast**, at least one port is handling an active call and at least one port is out of service.

- **Off:** No ports are assigned.

Monitor and Telco LEDs

The Monitor and Telco LEDs indicate line coding, network framing, and loopback status. These LEDs are color coded—green, yellow, and red. The Monitor and Telco LED descriptions follow.

Telco and Monitor LED alarms and errors are logged as switch events in ShoreTel Director's event log.

- **Line Coding:** This LED indicates line coding status, as follows:
  — **When green**, the line coding signal is good.
  — **When yellow**, bipolar violations are being received at one-second intervals.
  — **When red**, a loss of signal (LOS) has occurred.
  — **When flashing red**, loopback is active (local or CO).
  — **When off**, the switch has no power.
- **Framing:** This LED indicates network framing status, as follows:
  — **When green**, the T1/E1 signal is *in frame*; the signal is synchronized.
  — **When yellow**, a yellow alarm has been received from the Central Office.
  — **When flashing yellow**, the frame-bit error rate has exceeded its limits. A small number of frame-bit errors (>1 per million) have occurred; this state will take up to 10 minutes to clear.
  — **When flashing fast yellow**, a series of frame-bit errors (>1 per 1000) occurred.
  — **When red**, the T1/E1 signal is out-of-frame (OOF). The received signal cannot be framed to the Extended Superframe (ESF) or D4 format.
  — **When flashing red**, loopback is active (local or CO).
  — **When off**, the switch has no power.

C.2.6.3 ShoreTel Voice Switch 220T1 Connectors

The ShoreTel Voice Switch 220T1 voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-45 T1 telco port
- 1 RJ-45 T1 monitor port for connecting test equipment

C.2.7 ShoreTel Voice Switch 220T1A

The following sections describe ShoreTel Voice Switch 220T1A resource capacity, LED behavior, and connectors. The ShoreTel Voice Switch 220T1A is not supported in installations outside the U.S. and Canada. Figure C-8 displays the ShoreTel Voice Switch 220T1A front plate.

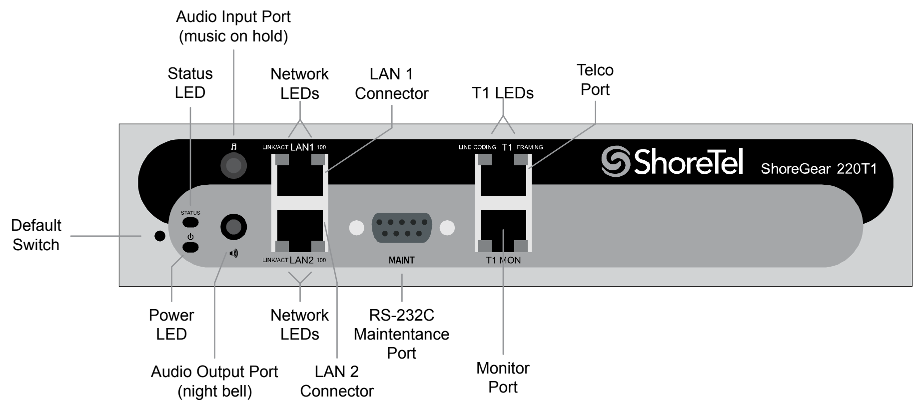


Figure C-8 ShoreTel Voice Switch 220T1A Front Plate

C.2.7.1 Switch Capacity

- **Analog Circuit Resources**
  - **Ports 1-2:** Two Loop Start Trunks
  - **Ports 9-12:** Four Extensions or DID Trunks. A single command configures all ports as either Extensions or DID trunks.
  - **Power Failure Transfer Unit:** Trunk Port 1 to Extension Port 12
- **Digital Circuit Resources:** 24 channels maximum
  - **One T1 circuit, 24 channels per circuit**
- **Make Me Conference Resource:** Six ports
  - **Ports 1-2, 9-12**
- **Maximum IP Phone Resources:** 220 devices
  - **Analog Channel Reallocation:** 30
  - **Digital Channel Reallocation:** 120
  - **Built-in Resources:** 70

C.2.7.2 LED Descriptions

Power LED

The ShoreTel Voice Switch 220T1A has one power LED, which indicates the following:

- **On:** The switch is operating normally.
- **Off:** The switch has no power.
- **Flashing**

— **2 flashes—**The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.

— **3 flashes—**Booting via FTP. Flash memory might be corrupted. Go to the Quick Look page to ensure that the system is running properly.

— **4 flashes—**The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.

— **5 flashes—**The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreTel server, and set the boot file to `/tsk/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreTel server's IP address, and set option 67 to `/tsk/vxworks`.

— **6 flashes—**Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

The ShoreTel Voice Switch 220T1A network LEDs (LAN1 and LAN2) show network activity and indicate the speed at which the switch is communicating with the network.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:

  — **When flashing**, network activity is detected.

  — **When on (not flashing)**, the switch is connected to an Ethernet network.

  — **When off**, the switch cannot detect an Ethernet network.

  This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- **100M**

  — **When green**, the switch is connected to a 100BaseT network.

  — **When off**, the switch is connected to a 10BaseT network.

Status LED

The ShoreTel Voice Switch 220T1A has one status LED to provide general information about the ports. The color and blink pattern of the LED indicate the port function:

- **Status LED (Green)**
  - **When on steady**, no ports are handling active calls.
  - **When flashing fast (100 msec on/off)**, at least one port is handling an active call.
- **Status LED (Yellow)**
  - **When on steady**, no ports are handling active calls and at least one port is out of service.
  - **When flashing slow (1 sec. on/off)**, the switch is not connected (or has lost connection) to a ShoreTel server.
  - **When flashing fast (100 msec on/off)**, at least one port is handling an active call and at least one port is out of service.
- **Off:** No ports are assigned.

Monitor and Telco LEDs

The Monitor and Telco LEDs indicate line coding, network framing, and loopback status. These LEDs are color coded—green, yellow, and red. The Monitor and Telco LED descriptions follow.

Telco and Monitor LED alarms and errors are logged as switch events in ShoreTel Director's event log.

- **Line Coding:** This LED indicates line coding status, as follows:
  - **When green**, the line coding signal is good.
  - **When yellow**, bipolar violations (BPV) are received at one second intervals.
  - **When red**, a loss of signal (LOS) has occurred.
  - **When flashing red**, loopback is active (local or CO).
  - **When off**, the switch has no power.
- **Framing:** This LED indicates network framing status, as follows:
  - **When green**, the T1/E1 signal is *in frame*; the signal is synchronized.
  - **When yellow**, a yellow alarm has been received from the Central Office.
  - **When flashing yellow**, the frame-bit error rate has exceeded its limits.
  - **When flashing slow yellow**, a small number of frame-bit errors (>1 per million) have occurred; this state will take up to 10 minutes to clear.
  - **When flashing fast yellow**, a series of frame-bit errors (> 1 per 1000) have occurred.
  - **When red**, the T1/E1 signal is out-of-frame (OOF). The received signal cannot be framed to the Extended Superframe (ESF) or D4 format.
  - **When flashing red,** loopback is active (local or CO).
  - **When off**, the switch has no power.

C.2.7.3 ShoreTel Voice Switch 220T1A Connectors

The ShoreTel Voice Switch 220T1A voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 1 RJ-21X male connector for mass termination of the telephone/trunk ports
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-45 T1 telco port
- 1 RJ-45 T1 monitor port for connecting test equipment

ShoreTel Voice Switch 220T1A RJ-21X Telephone and Trunk Connector

Table C-6 lists the RJ-21X Ring and Tip pin numbers for the SG 220T1A.

Table C-6 ShoreTel Voice Switch 220T1A RJ-21X Telephone and Trunk Connector Pins

		Ring		Tip	
Port	Type	Pin #	Cable Color	Pin #	Cable Color
1	Trunk	1	Blue/White	26	White/Blue
–		2	Orange/White	27	White/Orange
2	Trunk	3	Green/White	28	White/Green
–		4	Brown/White	29	White/Brown
–		5	Slate/White	30	White/Slate
–		6	Blue/Red	31	Red/Blue
–		7	Orange/Red	32	Red/Orange
–		8	Green/Red	33	Red/Green
–		9	Brown/Red	34	Red/Brown
–		10	Slate/Red	35	Red/Slate
–		11	Blue/Black	36	Black/Blue
–		12	Orange/Black	37	Black/Orange
–		13	Green/Black	38	Black/Green
–		14	Brown/Black	39	Black/Brown
–		15	Slate/Black	40	Black/Slate
–		16	Blue/Yellow	41	Yellow/Blue
9	Extension	17	Orange/Yellow	42	Yellow/Orange
–		18	Green/Yellow	43	Yellow/Green
10	Extension	19	Brown/Yellow	44	Yellow/Brown
–		20	Slate/Yellow	45	Yellow/Slate
11	Extension	21	Blue/Violet	46	Violet/Blue

Table C-6 ShoreTel Voice Switch 220T1A RJ-21X Telephone and Trunk Connector Pins (Continued)

		Ring		Tip	
Port	Type	Pin #	Cable Color	Pin #	Cable Color
–		22	Orange/Violet	47	Violet/Orange
12	Extension	23	Green/Violet	48	Violet/Green
–		24	Brown/Violet	49	Violet/Brown
–		25	Slate/Violet	50	Violet/Slate

C.2.8 ShoreTel Voice Switch 220E1

The following sections describe ShoreTel Voice Switch 220E1 resource capacity, LED behavior, and connectors. Figure C-9 displays the ShoreTel Voice Switch 220E1 front plate.

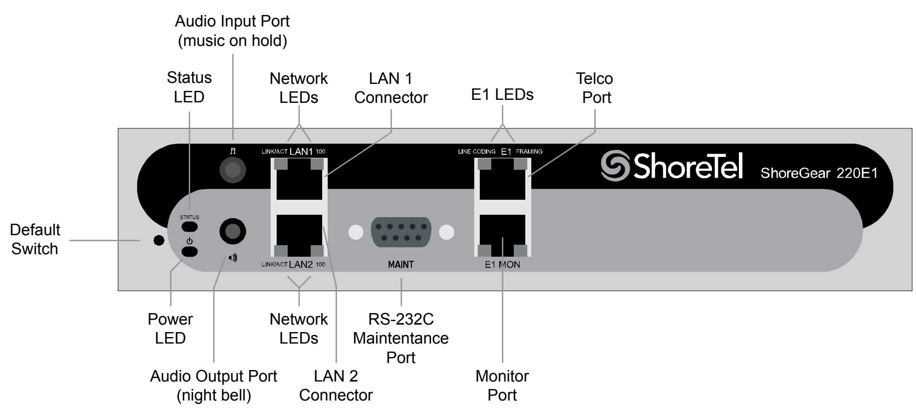


Figure C-9 ShoreTel Voice Switch 220E1 Front Plate

C.2.8.1 Switch Capacity

- **Digital Circuit Resources**: 30 channels maximum
  - **One E1 circuit, 30 channels per circuit**
- **Make Me Conference Resource**: none
- **Maximum IP Phone Resources**: 220
  - **Digital Channel Reallocation**: 150
  - **Built-in Resources**: 70

C.2.8.2 LED Descriptions

Power LED

The ShoreTel Voice Switch 220E1 has one power LED, which indicates the following:

- **On**: The switch is operating normally.
- **Off**: The switch has no power.
- **Flashing**

— **2 flashes—**The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.

— **3 flashes—**Booting via FTP. Flash memory might be corrupted. Go to the Quick Look page to ensure that the system is running properly.

— **4 flashes—**The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.

— **5 flashes—**The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreTel server, and set the boot file to `/tsk/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreTel server's IP address, and set option 67 to `/tsk/vxworks`.

— **6 flashes—**Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

The ShoreTel Voice Switch 220E1 network LEDs (LAN1 and LAN2) indicate network activity and the speed at which the switch is communicating with the network.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:

  — **When flashing**, network activity is detected.

  — **When on (not flashing)**, the switch is connected to an Ethernet network.

  — **When off**, the switch cannot detect an Ethernet network.

  This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- **100M**

  — **When green**, the switch is connected to a 100BaseT network.

  — **When off**, the switch is connected to a 10BaseT network.

Status LED

The ShoreTel Voice Switch 220E1 has one status LED to provide general information about the ports. The color and blink pattern of the LED indicate the port function:

- **Status LED (Green)**
  - **When on steady**, no ports are handling active calls.
  - **When flashing fast**, at least one port is handling an active call.
- **Status LED (Yellow)**
  - **When on steady**, no ports are handling active calls and at least one port is out of service.
  - **When flashing slow**, the switch is not connected (or has lost connection) to a ShoreTel server.
  - **When flashing fast**, at least one port is handling an active call and at least one port is out of service.
- **Off:** No ports are assigned.

Monitor and Telco LEDs

The Monitor and Telco LEDs indicate line coding, network framing, and loopback status. These LEDs are color coded—green, yellow, and red. The Monitor and Telco LED descriptions follow.

Telco and Monitor LED alarms and errors are logged as switch events in ShoreTel Director's event log.

- **Line Coding:** This LED indicates line coding status, as follows:
  - **When green**, the line coding signal is good.
  - **When yellow**, bipolar violations (BPV) are being received at one second intervals.
  - **When red**, a loss of signal (LOS) has occurred.
  - **When flashing red**, loopback is active (local or CO).
  - **When off**, the switch has no power.
- **Framing:** This LED indicates network framing status, as follows:
  - **When green**, the T1/E1 signal is *in frame*; the signal is synchronized.
  - **When yellow**, a yellow alarm has been received from the Central Office.
  - **When flashing yellow**, the frame-bit error rate has exceeded its limits.
  - **When flashing slow yellow**, a small number of frame-bit errors ($10e^{-6}$) have occurred; this state will take up to 10 minutes to clear.
  - **When flashing fast yellow**, a series of frame-bit errors ($10e^{-3}$) have occurred.
  - **When red**, the T1/E1 signal is out-of-frame (OOF). The received signal cannot be framed to the Extended Superframe (ESF) or D4 format.
  - **When flashing red**, loopback is active (local or CO).
  - **When off**, the switch has no power.

C.2.8.3 ShoreTel Voice Switch 220E1 Connectors

The ShoreTel Voice Switch 220E1 voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-45 T1 telco port
- 1 RJ-45 T1 monitor port for connecting test equipment

C.2.9 ShoreTel Voice Switch T1k

This section describe ShoreTel Voice Switch T1k resource capacity, LED behavior, and connectors. The only countries where the ShoreTel Voice Switch T1k is supported are the U.S. and Canada. Figure C-10 displays the ShoreTel Voice Switch T1k front plate.

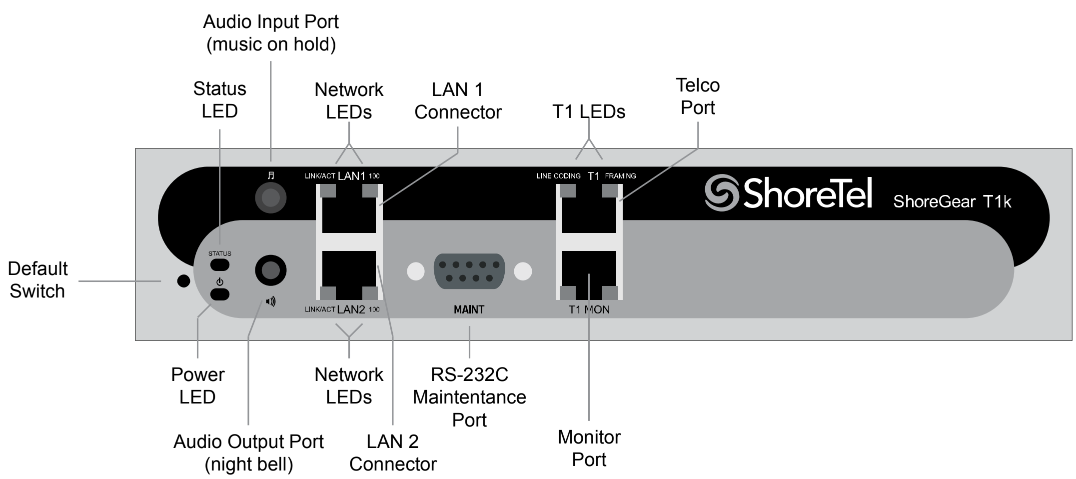


Figure C-10 ShoreTel Voice Switch T1k Front Plate

C.2.9.1 Switch Capacity

- **Digital Circuit Resources**: 24 channels maximum
  - **One T1 circuit, 24 channels per circuit**
- **Make Me Conference Resource**: None
- **Maximum IP Phone Resources**: None

C.2.9.2 LED Descriptions

Power LED

The ShoreTel Voice Switch T1k has one power LED, which indicates the following:

- **On**: The switch is operating normally.
- **Off**: The switch has no power.

- **Flashing**
  - — **2 flashes—**The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
  - — **3 flashes—**Booting via FTP. Flash memory might be corrupted. Go to the Quick Look page to ensure that the system is running properly.
  - — **4 flashes—**The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.
  - — **5 flashes—**The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreTel server, and set the boot file to `/tsk/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreTel server's IP address, and set option 67 to `/tsk/vxworks`.
  - — **6 flashes—**Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

The ShoreTel Voice Switch T1k network LEDs (LAN1 and LAN2) indicates that the network is active and the speed at which the switch is communicating with the network.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:
  - — **When flashing**, network activity is detected.
  - — **When on (not flashing)**, the switch is connected to an Ethernet network.
  - — **When off**, the switch cannot detect an Ethernet network.

  This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.
- **100M**
  - — **When green**, the switch is connected to a 100BaseT network.
  - — **When off**, the switch is connected to a 10BaseT network.

Status LED

The ShoreTel Voice Switch T1k has one status LED to provide general information about the ports. The color and blink pattern of the LED indicate the port function:

- **Status LED (Green)**
  - — **When on steady**, no ports are handling active calls.
  - — **When flashing fast**, at least one port is handling an active call.
- **Status LED (Yellow)**
  - — **When on steady**, no ports have active calls, and at least one port is out of service.
  - — **When flashing slow**, the switch is not connected (or has lost connection) to a ShoreTel server.
  - — **When flashing fast**, at least one port is handling an active call and at least one port is out of service.
- **Off:** No ports are assigned.

Monitor and Telco LEDs

The Monitor and Telco LEDs indicate line coding, network framing, and loopback status. These LEDs are color coded—green, yellow, and red. The Monitor and Telco LED descriptions follow.

Telco and Monitor LED alarms and errors are logged as switch events in ShoreTel Director's event log.

- **Line Coding:** This LED indicates line coding status, as follows:
  - — **When green**, the line coding signal is good.
  - — **When yellow**, bipolar violations (BPV) are being received at one second intervals.
  - — **When red**, a loss of signal (LOS) has occurred.
  - — **When flashing red**, loopback is active (local or CO).
  - — **When off**, the switch has no power.
- **Framing:** This LED indicates network framing status, as follows:
  - — **When green**, the T1/E1 signal is *in frame*; the signal is synchronized.
  - — **When yellow**, a yellow alarm has been received from the Central Office.
  - — **When flashing yellow**, the frame-bit error rate has exceeded its limits. A small number of frame-bit errors (>1 per million) have occurred; this state will take up to 10 minutes to clear.
  - — **When flashing fast yellow**, a series of frame-bit errors (>1 per 1000) have occurred.
  - — **When red**, the T1/E1 signal is out-of-frame (OOF). The received signal cannot be framed to the Extended Superframe (ESF) or D4 format.
  - — **When flashing red**, loopback is active (local or CO).
  - — **When off**, the switch has no power.

C.2.9.3 ShoreTel Voice Switch T1k Connectors

The ShoreTel Voice Switch T1k voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-45 T1 telco port
- 1 RJ-45 T1 monitor port for connecting test equipment

C.2.10 ShoreTel Voice Switch E1k

This section describes ShoreTel Voice Switch E1k resource capacity, LED behavior, and connectors. The only countries where the ShoreTel Voice Switch E1k is supported are the U.S. and Canada. Figure C-11 displays the ShoreTel Voice Switch E1k front plate.

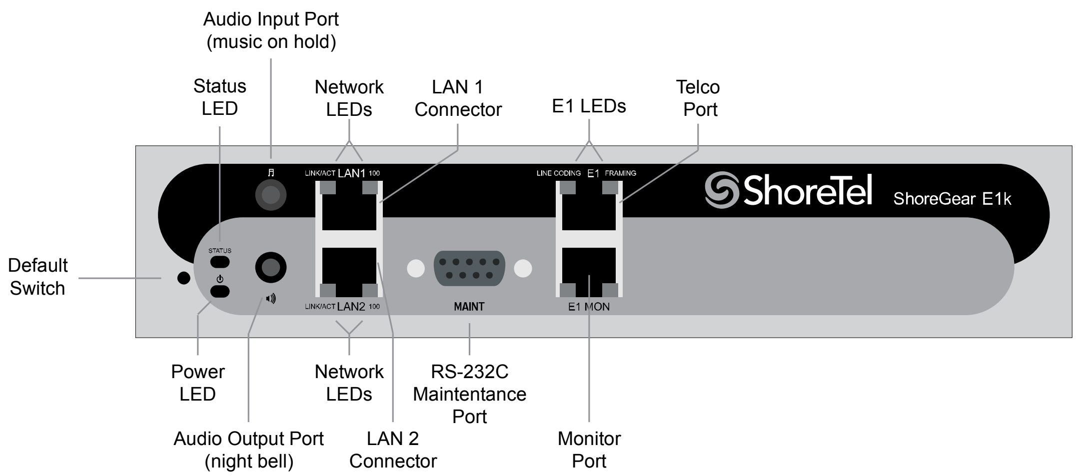


Figure C-11 ShoreTel Voice Switch E1k Front Plate

C.2.10.1 Switch Capacity

- **Digital Circuit Resources:** 30 channels maximum
  - **One E1 circuit, 30 channels per circuit**
- **Make Me Conference Resource:** None
- **Maximum IP Phone Resources:** None

C.2.10.2 LED Descriptions

Power LED

The ShoreTel Voice Switch E1k has one power LED, which indicates the following:

- **On:** The switch is operating normally.
- **Off:** The switch has no power.

- **Flashing**
  - **2 flashes—**The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
  - **3 flashes—**Booting via FTP. Flash memory might be corrupted. Go to the Quick Look page to ensure that the system is running properly.
  - **4 flashes—**The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.
  - **5 flashes—**The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreTel server, and set the boot file to `/tsk/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreTel server's IP address, and set option 67 to `/tsk/vxworks`.
  - **6 flashes—**Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

The ShoreTel Voice Switch E1k network LEDs (LAN1 and LAN2) show network activity and indicate the speed at which the switch is communicating with the network.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:
  - **When flashing**, network activity is detected.
  - **When on (not flashing)**, the switch is connected to an Ethernet network.
  - **When off**, the switch cannot detect an Ethernet network.

  This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.
- **100M**
  - **When green**, the switch is connected to a 100BaseT network.
  - **When off**, the switch is connected to a 10BaseT network.

Status LED

The ShoreTel Voice Switch E1k has one status LED to provide general information about the ports. The color and blink pattern of the LED indicate the port function:

- **Status LED (Green)**
  - **When on steady**, no ports are handling active calls.
  - **When flashing fast**, at least one port is handling an active call.
- **Status LED (Yellow)**
  - **When on steady**, no ports have active calls, and at least one port is out of service.
  - **When flashing slow**, the switch is not connected (or has lost connection) to a ShoreTel server.
  - **When flashing fast**, at least one port is handling an active call and at least one port is out of service.
- **Off:** No ports are assigned.

Monitor and Telco LEDs

The Monitor and Telco LEDs indicate line coding, network framing, and loopback status. These LEDs are color coded—green, yellow, and red. The Monitor and Telco LED descriptions follow.

Telco and Monitor LED alarms and errors are logged as switch events in ShoreTel Director's event log.

- **Line Coding:** This LED indicates line coding status, as follows:
  - **When green**, the line coding signal is good.
  - **When yellow**, bipolar violations (BPV) are being received at one second intervals.
  - **When red**, a loss of signal (LOS) has occurred.
  - **When flashing red**, loopback is active (local or CO).
  - **When off**, the switch has no power.
- **Framing:** This LED indicates network framing status, as follows:
  - **When green**, the E1 signal is *in frame*; the signal is synchronized.
  - **When yellow**, a yellow alarm has been received from the Central Office.
  - **When flashing yellow**, the frame-bit error rate has exceeded its limits. A small number of frame-bit errors (>1 per million) have occurred; this state will take up to 10 minutes to clear.
  - **When flashing fast yellow**, a series of frame-bit errors (>1 per 1000) have occurred.
  - **When red**, the T1/E1 signal is out-of-frame (OOF). The received signal cannot be framed to the Extended Superframe (ESF) or D4 format.
  - **When flashing red**, loopback is active (local or CO).
  - **When off**, the switch has no power.

C.2.10.3 ShoreTel Voice Switch E1k Connectors

The ShoreTel Voice Switch E1k voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-45 T1 telco port
- 1 RJ-45 T1 monitor port for connecting test equipment

C.3 Specifications for ShoreTel Voice Model Switches

C.3.1 ShoreTel Voice Switch 90V

This section describes ShoreTel Voice Switch 90V resource capacity, LED behavior, and connectors. The ShoreTel Voice Switch 90V is not supported in installations outside the U.S. and Canada. Figure C-12 displays the ShoreTel Voice Switch 90V front plate.

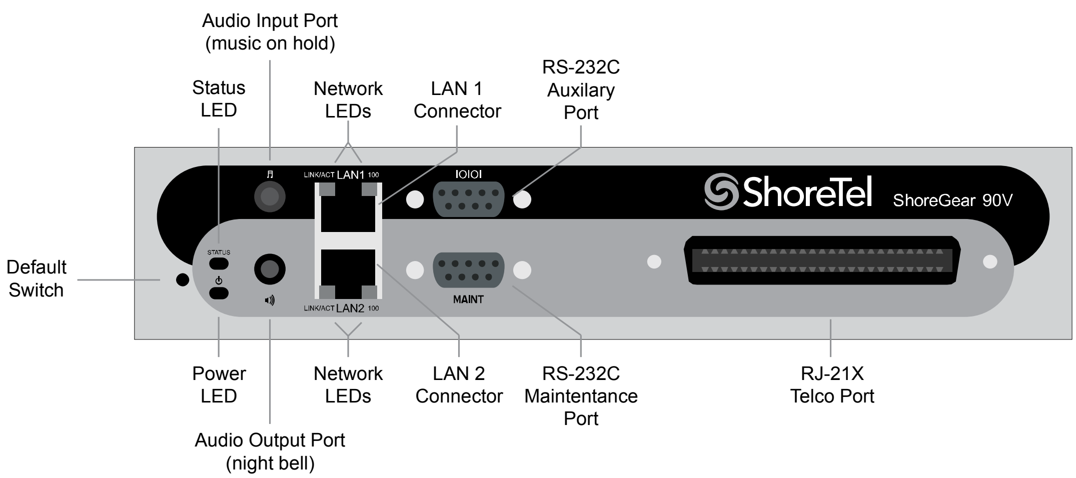


Figure C-12 ShoreTel Voice Switch 90V Front Plate

C.3.1.1 Switch Capacity

- Analog Circuit Resources
  - **Ports 1-8:** Eight Loop Start Trunks
  - **Ports 9-12:** Four Extensions or DID Trunks. A single command configures all ports as either Extensions or DID trunks.
  - **Power Failure Transfer Unit:** Trunk Port 1 to Extension Port 12
- **Make Me Conference Resources:** 12 ports
  - **Ports 1-12**
- **Maximum IP Phone Resources:** 90 devices
  - **Analog Port Reallocation:** 60
  - **Built-in Resources:** 30

C.3.1.2 LED Descriptions

Power LED

The ShoreTel Voice Switch 90V has one power LED, which indicates the following:

- **On:** The switch is operating normally.
- **Off:** The switch has no power.

- **Flashing:**
  - — **2 flashes—**The switch failed its internal self-test. This indicates a hardware failure. Replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
  - — **3 flashes—**Booting via FTP. Flash memory might be corrupted. Go to the Quick Look page to ensure that the system is running properly.
  - — **4 flashes—**The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.
  - — **5 flashes—**The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreTel server, and set the boot file to `/tsk/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreTel server's IP address, and set option 67 to `/tsk/vxworks`.
  - — **6 flashes—**Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

The ShoreTel Voice Switch 90V network LEDs (LAN1 and LAN2) shows network activity and indicate the speed at which the switch is communicating with the network.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:
  - — **When flashing**, network activity is detected.
  - — **When on (not flashing)**, the switch is connected to an Ethernet network.
  - — **When off**, the switch cannot detect an Ethernet network.

  This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.
- **100M:**
  - — **When green**, the switch is connected to a 100BaseT network.
  - — **When off**, the switch is connected to a 10BaseT network.

Status LED

The ShoreTel Voice Switch 90V has one status LED to provide general information about the ports. The color and blink pattern of the LED indicate the port function:

- **Status LED (Green)**
  - — **When on steady**, no ports are handling active calls.
  - — **When flashing fast**, at least one port is handling an active call.
- **Status LED (Yellow)**
  - — **When on steady**, no ports are handling active calls and at least one port is out of service.
  - — **When flashing slow**, the switch is not connected (or has lost connection) to a ShoreTel server.
  - — **When flashing fast**, at least one port is handling an active call and at least one port is out of service.
- **Off:** No ports are assigned.

C.3.1.3 ShoreTel Voice Switch 90V Connectors

The ShoreTel Voice Switch 90V voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-21X male connector for mass termination of the telephone/trunk ports
  - — **Power Failure Transfer Unit:** Trunk Port 1 to Extension Port 12
  - — **Backup Operator:** Extension Port 12

ShoreTel Voice Switch 90V RJ-21X Telephone and Trunk Connector

Table C-7 lists the RJ-21X Ring and Tip pin numbers for the ShoreTel Voice Switch 90V

Table C-7 ShoreTel Voice Switch 90V RJ-21X Telephone and Trunk Connector Pins

		Ring		Tip	
Port	Type	Pin #	Cable Color	Pin #	Cable Color
1	Trunk	1	Blue/White	26	White/Blue
–		2	Orange/White	27	White/Orange
2	Trunk	3	Green/White	28	White/Green
–		4	Brown/White	29	White/Brown
3	Trunk	5	Slate/White	30	White/Slate
–		6	Blue/Red	31	Red/Blue
4	Trunk	7	Orange/Red	32	Red/Orange

Table C-7 ShoreTel Voice Switch 90V RJ-21X Telephone and Trunk Connector Pins (Continued)

		Ring		Tip	
Port	**Type**	**Pin #**	**Cable Color**	**Pin #**	**Cable Color**
–		8	Green/Red	33	Red/Green
5	Trunk	9	Brown/Red	34	Red/Brown
–		10	Slate/Red	35	Red/Slate
6	Trunk	11	Blue/Black	36	Black/Blue
–		12	Orange/Black	37	Black/Orange
7	Trunk	13	Green/Black	38	Black/Green
–		14	Brown/Black	39	Black/Brown
8	Trunk	15	Slate/Black	40	Black/Slate
–		16	Blue/Yellow	41	Yellow/Blue
9	Extension - DID	17	Orange/Yellow	42	Yellow/Orange
–		18	Green/Yellow	43	Yellow/Green
10	Extension - DID	19	Brown/Yellow	44	Yellow/Brown
–		20	Slate/Yellow	45	Yellow/Slate
11	Extension - DID	21	Blue/Violet	46	Violet/Blue
–		22	Orange/Violet	47	Violet/Orange
12	Extension - DID	23	Green/Violet	48	Violet/Green
–		24	Brown/Violet	49	Violet/Brown
–		25	Slate/Violet	50	Violet/Slate

C.3.2 ShoreTel Voice Switch 90BRIV

This section describes ShoreTel Voice Switch 90BRIV resource capacity, LED behavior, and connectors. Figure C-13 displays the ShoreTel Voice Switch 90BRIV front plate.

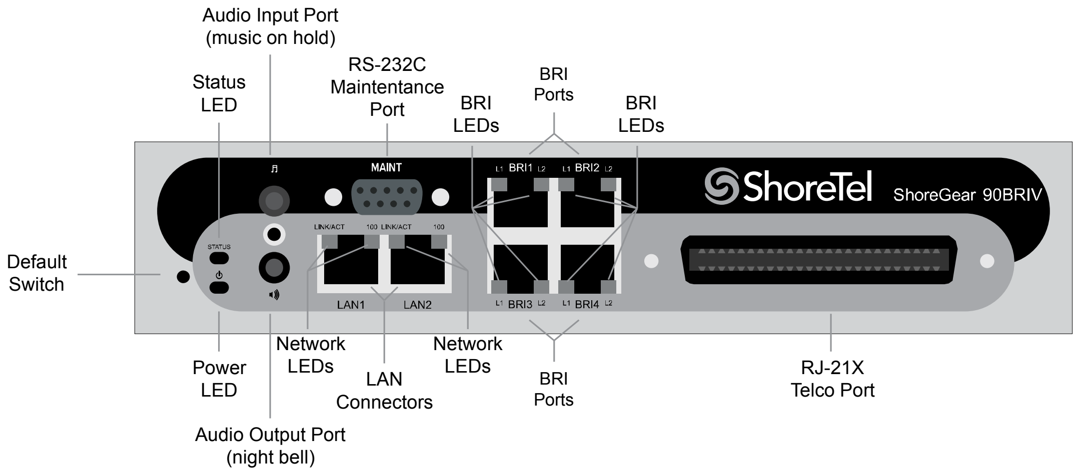


Figure C-13 ShoreTel Voice Switch 90BRIV Front Plate

C.3.2.1 Switch Capacity

- **Analog Circuit Resources**
  - **Ports 9-12**: Extensions
- **Digital Circuit Resources**
  - **Four BRI Spans, each comprising two channels**: Eight channels maximum
- **Make Me Conference Resources:** 4 ports
  - **Ports 9-12**
- **Maximum IP Phone Resources**: 90 devices
  - **Analog Port Reallocation:** 20
  - **Digital Channel Reallocation**: 40
  - **Built-in Resources**: 30

C.3.2.2 LED Descriptions

Power LED

The ShoreTel Voice Switch 90BRIV has one power LED, which indicates the following:

- **On**: The switch is operating normally.
- **Off**: The switch has no power.
- **Flashing**
  - **2 flashes—**The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
  - **3 flashes—**Booting via FTP. Flash memory might be corrupted. Go to the Quick Look page to ensure that the system is running properly.

— **4 flashes—**The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.

— **5 flashes—**The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreTel server, and set the boot file to `/tsk/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreTel server's IP address, and set option 67 to `/tsk/vxworks`.

— **6 flashes—**Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

The ShoreTel Voice Switch 90BRIV network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:

  — **When flashing**, network activity is detected.

  — **When on (not flashing)**, the switch is connected to an Ethernet network.

  — **When off**, the switch cannot detect an Ethernet network.

  This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- **100M**

  — **When green**, the switch is connected to a 100BaseT network.

  — **When off**, the switch is connected to a 10BaseT network.

Status LED

The ShoreTel Voice Switch 90BRIV has one status LED to provide general information about the ports. The color and blink pattern of the LED indicate the port function:

- **Status LED (Green)**

  — **When on steady**, no ports are handling active calls.

    — **When flashing fast (100 msec on/off)**, at least one port is handling an active call.

- **Status LED (Yellow)**
    — **When on steady**, no ports are handling active calls and at least one port is out of service.
    — **When flashing slow (1 sec. on/off)**, the switch is not connected (or has lost connection) to a ShoreTel server.
    — **When flashing fast (100 msec on/off)**, at least one port is handling an active call and at least one port is out of service.
- **Off**: No ports are assigned.

BRI LED

Each BRI connector has two LEDs to indicate port activity. The color and blink pattern of the LED indicate the port function:

- **LED 1: Off, LED 2 Off –** Port not configured in Director
- **LED 1: Yellow, LED 2 Off –** Port inactive or not connected
- **LED 1: Off, LED 2 Off –** Layer 1 active. Layer 2 not established
- **LED 1: Off, LED 2 Green –** Layer 1 active. Layer 2 active.
- **LED 1: Off, LED 2 Green flashing –** Call in progress (Layer 1, Layer 2, and Layer 3 active).

C.3.2.3 ShoreTel Voice Switch 90BRIV Connectors

The ShoreTel Voice Switch 90BRIV voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-21X male connector for mass termination of the telephone/trunk ports
- 4 RJ-45 T1 telco port

ShoreTel Voice Switch 90BRIV RJ-21X Telephone and Trunk Connector

Table C-8 lists the RJ-21X Ring and Tip pin numbers for the SG 90BRIV.

Table C-8 ShoreTel Voice Switch 90BRIV RJ-21X Telephone and Trunk Connector Pins

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
–		1	Blue/White	26	White/Blue
–		2	Orange/White	27	White/Orange
–		3	Green/White	28	White/Green
–		4	Brown/White	29	White/Brown

Table C-8 ShoreTel Voice Switch 90BRIV RJ-21X Telephone and Trunk Connector Pins (Continued)

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
–		5	Slate/White	30	White/Slate
–		6	Blue/Red	31	Red/Blue
–		7	Orange/Red	32	Red/Orange
–		8	Green/Red	33	Red/Green
–		9	Brown/Red	34	Red/Brown
–		10	Slate/Red	35	Red/Slate
–		11	Blue/Black	36	Black/Blue
–		12	Orange/Black	37	Black/Orange
–		13	Green/Black	38	Black/Green
–		14	Brown/Black	39	Black/Brown
–		15	Slate/Black	40	Black/Slate
–		16	Blue/Yellow	41	Yellow/Blue
9	Extension	17	Orange/Yellow	42	Yellow/Orange
–		18	Green/Yellow	43	Yellow/Green
10	Extension	19	Brown/Yellow	44	Yellow/Brown
–		20	Slate/Yellow	45	Yellow/Slate
11	Extension	21	Blue/Violet	46	Violet/Blue
–		22	Orange/Violet	47	Violet/Orange
12	Extension	23	Green/Violet	48	Violet/Green
–		24	Brown/Violet	49	Violet/Brown
–		25	Slate/Violet	50	Violet/Slate

C.3.3 ShoreTel Voice Switch 50V

This section describes ShoreTel Voice Switch 50V resource capacity, LED behavior, and connectors. The ShoreTel Voice Switch 50V is not supported in installations outside the U.S. and Canada. Figure C-14 displays the ShoreTel Voice Switch 50V front plate.

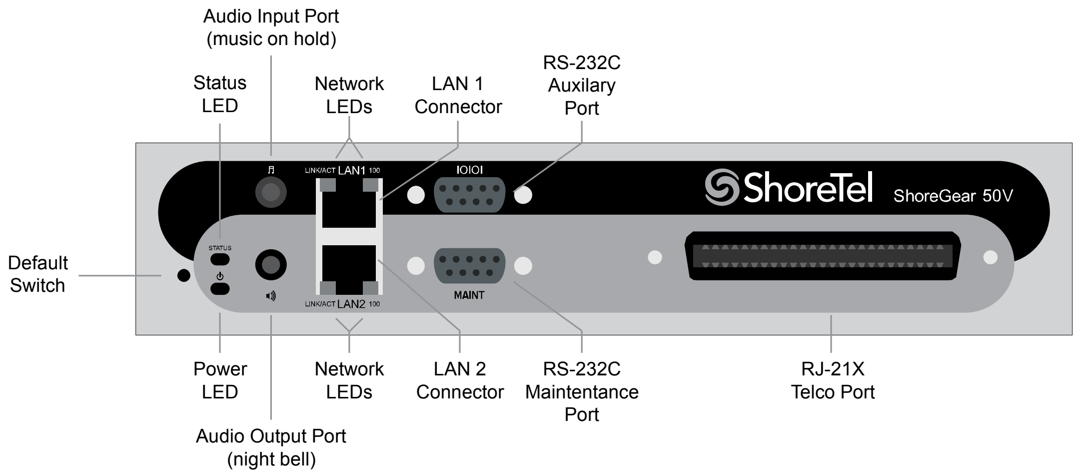


Figure C-14 ShoreTel Voice Switch 50V Front Plate

C.3.3.1 Switch Capacity

- **Analog Circuit Resources**
  - **Ports 1-4:** Four Loop Start Trunks
  - **Ports 11-12:** Two Extensions or DID Trunks. A single command configures all ports as either Extensions or DID trunks.
  - **Power Failure Transfer Unit:** Trunk Port 1 to Extension Port 12
- **Make Me Conference Resources:** six ports
  - **Ports 1-4, 11-12**
- **Maximum IP Phone Resources:** 50 devices
  - **Analog Port Reallocation:** 30
  - **Built-in Resources:** 20

C.3.3.2 LED Descriptions

Power LED

The ShoreTel Voice Switch 50V has one power LED, which indicates the following:

- **On:** The switch is operating normally.
- **Off:** The switch has no power.
- **Flashing**
  - **2 flashes—**The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
  - **3 flashes—**Booting via FTP. Flash memory might be corrupted. Go to the Quick Look page to ensure that the system is running properly.
  - **4 flashes—**The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory

to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.

— **5 flashes—**The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreTel server, and set the boot file to `/tsk/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreTel server's IP address, and set option 67 to `/tsk/vxworks`.

— **6 flashes—**Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

The ShoreTel Voice Switch 50V network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:

  — **When flashing**, network activity is detected.

  — **When on (not flashing)**, the switch is connected to an Ethernet network.

  — **When off**, the switch cannot detect an Ethernet network.

  This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- **100M**

  — **When green**, the switch is connected to a 100BaseT network.

  — **When off**, the switch is connected to a 10BaseT network.

Status LED

The ShoreTel Voice Switch 50V has one status LED to provide general information about the ports. The color and blink pattern of the LED indicate the port function:

- **Status LED (Green)**

  — **When on steady**, no ports are handling active calls.

  — **When flashing fast**, at least one port is handling an active call.

- **Status LED (Yellow)**
  - **When on steady**, no ports are handling active calls and at least one port is out of service.
  - **When flashing slow**, the switch is not connected (or has lost connection) to a ShoreTel server.
  - **When flashing fast**, at least one port is handling an active call and at least one port is out of service.
- **Off:** No ports are assigned.

C.3.3.3 ShoreTel Voice Switch 50V Connectors

The ShoreTel Voice Switch 50V voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-21X male connector for mass termination of the telephone/trunk ports
  - **Power Failure Transfer Unit:** Trunk Port 1 to Extension Port 12
  - **Backup Operator:** Extension Port 12

ShoreTel Voice Switch 50V RJ-21X Telephone and Trunk Connector

Table C-9 lists the RJ-21X Ring and Tip pin numbers for the SG 50V.

Table C-9 ShoreTel Voice Switch 50V RJ-21X Telephone and Trunk Connector Pins

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
1	Trunk	1	Blue/White	26	White/Blue
–		2	Orange/White	27	White/Orange
2	Trunk	3	Green/White	28	White/Green
–		4	Brown/White	29	White/Brown
3	Trunk	5	Slate/White	30	White/Slate
–		6	Blue/Red	31	Red/Blue
4	Trunk	7	Orange/Red	32	Red/Orange
–		8	Green/Red	33	Red/Green
–		9	Brown/Red	34	Red/Brown
–		10	Slate/Red	35	Red/Slate
–		11	Blue/Black	36	Black/Blue
–		12	Orange/Black	37	Black/Orange
–		13	Green/Black	38	Black/Green

Table C-9 ShoreTel Voice Switch 50V RJ-21X Telephone and Trunk Connector Pins (Continued)

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
–		14	Brown/Black	39	Black/Brown
–		15	Slate/Black	40	Black/Slate
–		16	Blue/Yellow	41	Yellow/Blue
9	Extension - DID	17	Orange/Yellow	42	Yellow/Orange
–		18	Green/Yellow	43	Yellow/Green
10	Extension - DID	19	Brown/Yellow	44	Yellow/Brown
–		20	Slate/Yellow	45	Yellow/Slate
11	Extension - DID	21	Blue/Violet	46	Violet/Blue
–		22	Orange/Violet	47	Violet/Orange
12	Extension - DID	23	Green/Violet	48	Violet/Green
–		24	Brown/Violet	49	Violet/Brown
–		25	Slate/Violet	50	Violet/Slate

C.4 Specification for ShoreTel 1U Full-Width Voice Switches

C.4.1 ShoreTel Voice Switch 120

This section describes ShoreTel Voice Switch 120 resource capacity, LED behavior, and connectors. The ShoreTel Voice Switch 120 is not supported in installations outside the U.S. and Canada. Figure C-15 displays the ShoreTel Voice Switch 120 front plate.

The ShoreTel Voice Switch 120 is often referred to as the ShoreTel 120/24 (SG 120/24).

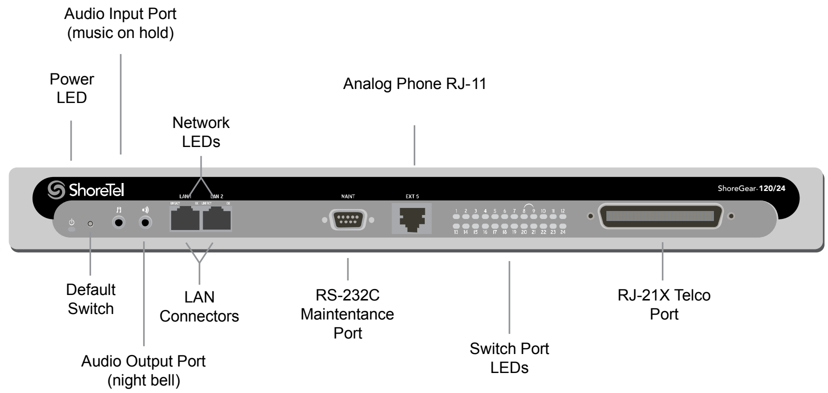


Figure C-15 ShoreTel Voice Switch 120 Front Plate

C.4.1.1 Switch Capacity

- **Analog Circuit Resources**
  - **Ports 1-8**: Eight Loop Start Trunks, DID Trunks, or Extensions
  - **Ports 9-24**: Sixteen Extensions.
  - **Power Failure Transfer Unit:** Trunk Port 8 to Extension Port 9
- **Make Me Conference Resource**: 24 Ports
  - **Ports 1-24**
- **Maximum IP Phone Resources**: 120 devices
  - **Analog Port Reallocation:** 120

C.4.1.2 LED Descriptions

Power LED

The ShoreTel Voice Switch 120 has one power LED, which indicates the following:

- **On**: The switch is operating normally.
- **Off**: The switch has no power.
- **Flashing**:
  - **2 flashes—**The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
  - **3 flashes—**Booting via FTP. Flash memory might be corrupted. Go to the Quick Look page to ensure that the system is running properly.
  - **4 flashes—**The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.

— **5 flashes—**The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreTel server, and set the boot file to `/tsa/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreTel server's IP address, and set option 67 to `/tsa/vxworks`.

— **6 flashes—**Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Switch Port LEDs

The ShoreTel Voice Switch 120 has 24 telephone/trunk port LEDs. The color of the LED indicates the port function:

- **Green** when the port is a telephone port.
- **Yellow** when the port is a trunk port.
- **Off** indicates the port is reserved for IP phones, for conferencing, or is unconfigured.

The following describes the switch port LED behavior and meaning:

- **Telephone Port LEDs (Green)**
  - **When on steady**, the port is configured as a telephone port and the telephone is idle.
  - **When flashing with ring cadence**, the telephone is ringing.
  - **When flashing slowly**, the telephone is off hook.
  - **When flashing fast**, the port is in use (call in progress).
- **Trunk Port LED (Yellow):**
  - **When on steady**, the port is configured as a trunk port and the trunk is idle.
  - **When flashing slowly**, the trunk is off hook.
  - **When flashing fast**, the trunk is in use (call in progress).
- **Port LED Alternating Green/Yellow:** The port is out of service. The LED periodically alternates green/yellow or yellow/green. The color of the LED between alternating colors indicates the port type: green for phone and yellow for trunk.
- **Off (IP phone):** When the LED is off, the port is reserved for IP phones, for conferencing, or is unconfigured.

Network LEDs

The network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:
  - — **When flashing**, network activity is detected.
  - — **When on (not flashing)**, heavy network activity is detected.
  - — **When off**, the switch has no power.

  This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.
- **100M**
  - — **When green**, the switch is connected to a 100BaseT network.
  - — **When off**, the switch is connected to a 10BaseT network.

C.4.1.3 ShoreTel Voice Switch 120 Connectors

The ShoreTel Voice Switch 120 voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-11 connector for connecting an analog phone (extension 9)
- 1 RJ-21X male connector for mass termination of the telephone/trunk ports
  - — **Power Failure Transfer Unit:** Trunk Port 8 to Extension Port 9
  - — **Backup Operator:** Extension Port 9

ShoreTel Voice Switch 120 RJ-21X Telephone and Trunk Connector

Table C-10 lists the RJ-21X Ring and Tip pin numbers for the SG 120I.

Table C-10 ShoreTel Voice Switch 120 RJ-21X Telephone and Trunk Connector Pins

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
1	Trunk, DID, Extension	1	Blue/White	26	White/Blue
2	Trunk, DID, Extension	2	Orange/White	27	White/Orange
3	Trunk, DID, Extension	3	Green/White	28	White/Green
4	Trunk, DID, Extension	4	Brown/White	29	White/Brown

Table C-10 ShoreTel Voice Switch 120 RJ-21X Telephone and Trunk Connector Pins (Continued)

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
5	Trunk, DID, Extension	5	Slate/White	30	White/Slate
6	Trunk, DID, Extension	6	Blue/Red	31	Red/Blue
7	Trunk, DID, Extension	7	Orange/Red	32	Red/Orange
8	Trunk, DID, Extension	8	Green/Red	33	Red/Green
9	Extension	9	Brown/Red	34	Red/Brown
10	Extension	10	Slate/Red	35	Red/Slate
11	Extension	11	Blue/Black	36	Black/Blue
12	Extension	12	Orange/Black	37	Black/Orange
13	Extension	13	Green/Black	38	Black/Green
14	Extension	14	Brown/Black	39	Black/Brown
15	Extension	15	Slate/Black	40	Black/Slate
16	Extension	16	Blue/Yellow	41	Yellow/Blue
17	Extension	17	Orange/Yellow	42	Yellow/Orange
18	Extension	18	Green/Yellow	43	Yellow/Green
19	Extension	19	Brown/Yellow	44	Yellow/Brown
20	Extension	20	Slate/Yellow	45	Yellow/Slate
21	Extension	21	Blue/Violet	46	Violet/Blue
22	Extension	22	Orange/Violet	47	Violet/Orange
23	Extension	23	Green/Violet	48	Violet/Green
24	Extension	24	Brown/Violet	49	Violet/Brown
–		25	Slate/Violet	50	Violet/Slate

C.4.2 ShoreTel Voice Switch 24A

This section describes ShoreTel Voice Switch 24A resource capacity, LED behavior, and connectors. The ShoreTel Voice Switch 24A is not supported in installations outside the U.S. and Canada. Figure C-16 displays the ShoreTel Voice Switch 24A front plate.

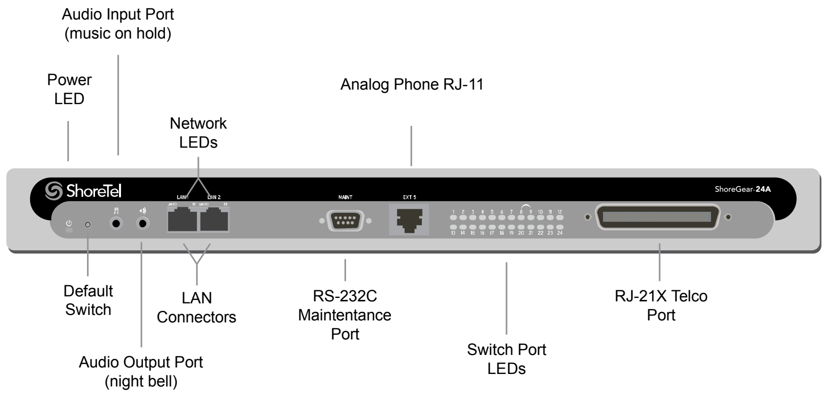


Figure C-16 ShoreTel Voice Switch 24A Front Plate

C.4.2.1 Switch Capacity

- **Analog Circuit Resources**
  - **Ports 1-24**: Twenty four extensions
- **Make Me Conference Resource**: 24 Ports
  - **Ports 1-24**
- **Maximum IP Phone Resources**: None

C.4.2.2 LED Descriptions

Power LED

The ShoreTel Voice Switch 24A has one power LED, which indicates the following:

- **On**: The switch is operating normally.
- **Off**: The switch has no power.
- **Flashing**:
  - **2 flashes—**The switch failed its internal self-test. Hardware failed; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
  - **3 flashes—**Booting via FTP. Flash memory might be corrupted. Go to the Quick Look page to ensure that the system is running properly.
  - **4 flashes—**The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.
  - **5 flashes—**The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreTel server, and set the boot file to `/tsa/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreTel server's IP address, and set option 67 to `/tsa/vxworks`.

— **6 flashes—**Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Switch Port LEDs

The ShoreTel Voice Switch 24A has 24 telephone/trunk port LEDs. The color of the LED indicates the port function:

- **Green** when the port is a telephone port.
- **Yellow** when the port is a trunk port.
- **Off** indicates the port is reserved for IP phones, for conferencing, or is unconfigured.

The following describes the switch port LED behavior and meaning:

- **Telephone Port LEDs (Green)**
  - **When on steady**, the port is configured as a telephone port and the telephone is idle.
  - **When flashing with ring cadence**, the telephone is ringing.
  - **When flashing slowly**, the telephone is off hook.
  - **When flashing fast**, the port is in use (call in progress).
- **Trunk Port LED (Yellow):**
  - **When on steady**, the port is configured as a trunk port and the trunk is idle.
  - **When flashing slowly**, the trunk is off hook.
  - **When flashing fast**, the trunk is in use (call in progress).
- **Port LED Alternating Green/Yellow:** The port is out of service. The LED periodically alternates green/yellow or yellow/green. The color of the LED between alternating colors indicates the port type: green for phone and yellow for trunk.
- **Off (IP phone):** When the LED is off, the port is reserved for IP phones, for conferencing, or is unconfigured.

Network LEDs

The network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:
  - **When flashing**, network activity is detected.
  - **When on (not flashing)**, heavy network activity is detected.

— **When off**, the switch has no power.

This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- **100M**
  - — **When green**, the switch is connected to a 100BaseT network.
  - — **When off**, the switch is connected to a 10BaseT network.

C.4.2.3 ShoreTel Voice Switch 24A Connectors

The ShoreTel Voice Switch 24A voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-11 connector for connecting an analog phone (extension 9)
- 1 RJ-21X male connector for mass termination of the telephone/trunk ports

ShoreTel Voice Switch 24A RJ-21X Telephone and Trunk Connector

Table C-11 lists the RJ-21X Ring and Tip pin numbers for the SG 24AI.

Table C-11 ShoreTel Voice Switch 24A RJ-21X Telephone and Trunk Connector Pins

		Ring		Tip	
Port	Type	Pin #	Cable Color	Pin #	Cable Color
1	Extension	1	Blue/White	26	White/Blue
2	Extension	2	Orange/White	27	White/Orange
3	Extension	3	Green/White	28	White/Green
4	Extension	4	Brown/White	29	White/Brown
5	Extension	5	Slate/White	30	White/Slate
6	Extension	6	Blue/Red	31	Red/Blue
7	Extension	7	Orange/Red	32	Red/Orange
8	Extension	8	Green/Red	33	Red/Green
9	Extension	9	Brown/Red	34	Red/Brown
10	Extension	10	Slate/Red	35	Red/Slate
11	Extension	11	Blue/Black	36	Black/Blue
12	Extension	12	Orange/Black	37	Black/Orange
13	Extension	13	Green/Black	38	Black/Green
14	Extension	14	Brown/Black	39	Black/Brown
15	Extension	15	Slate/Black	40	Black/Slate

Table C-11 ShoreTel Voice Switch 24A RJ-21X Telephone and Trunk Connector Pins (Continued)

Port	Type	Ring Pin #	Ring Cable Color	Tip Pin #	Tip Cable Color
16	Extension	16	Blue/Yellow	41	Yellow/Blue
17	Extension	17	Orange/Yellow	42	Yellow/Orange
18	Extension	18	Green/Yellow	43	Yellow/Green
19	Extension	19	Brown/Yellow	44	Yellow/Brown
20	Extension	20	Slate/Yellow	45	Yellow/Slate
21	Extension	21	Blue/Violet	46	Violet/Blue
22	Extension	22	Orange/Violet	47	Violet/Orange
23	Extension	23	Green/Violet	48	Violet/Green
24	Extension	24	Brown/Violet	49	Violet/Brown
–		25	Slate/Violet	50	Violet/Slate

C.4.3 ShoreTel Voice Switch 60

The following sections describe ShoreTel Voice Switch 60 resource capacity, LED behavior, and connectors. The ShoreTel Voice Switch 60 is not supported in installations outside the U.S. and Canada. Figure C-17 displays the ShoreTel Voice Switch 60 front plate.

The ShoreTel Voice Switch 60 is often referred to as the ShoreTel 60/12 (SG 60/12).

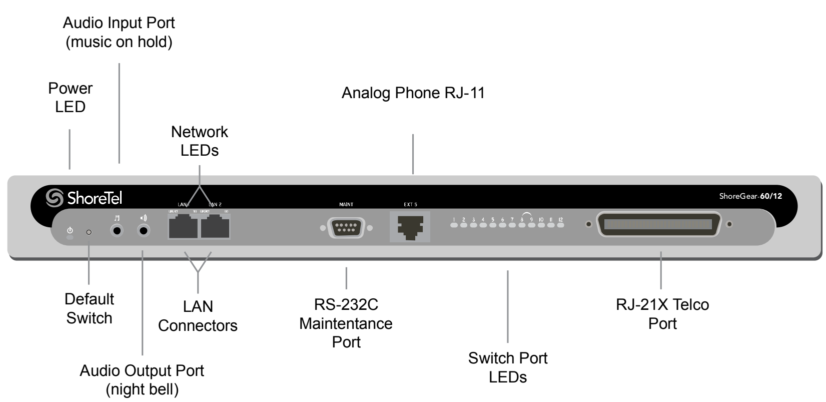


Figure C-17 ShoreTel Voice Switch 60 Front Plate

C.4.3.1 Switch Capacity

- **Analog Circuit Resources**
  - — **Ports 1-8**: Eight Loop Start Trunks, DID Trunks, or Extensions
  - — **Ports 9-12**: Four Extensions.
  - — **Backup Operator:** Extension Port 9
  - — **Power Failure Transfer Unit:** Trunk Port 8 to Extension Port 9

- **Make Me Conference Resource:** 12 ports
  - **Ports 1-12**
- **Maximum IP Phone Resources:** 60 devices
  - **Analog Port Reallocation:** 60

C.4.3.2 LED Descriptions

Power LED

The ShoreTel Voice Switch 60 has one power LED, which indicates the following:

- **On**: The switch is operating normally.
- **Off:** The switch has no power.
- **Flashing:**
  - **2 flashes—**The switch failed its internal self-test. Hardware has failed; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
  - **3 flashes—**Booting via FTP. Flash memory might be corrupted. Go to the Quick Look page to ensure that the system is running properly.
  - **4 flashes—**The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.
  - **5 flashes—**The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreTel server, and set the boot file to /tsa/vxworks. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreTel server's IP address, and set option 67 to /tsa/vxworks.
  - **6 flashes—**Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Switch Port LEDs

The ShoreTel Voice Switch 60 has 12 telephone/trunk port LEDs. The color of the LED indicates the port function:

- **Green** when the port is a telephone port.
- **Yellow** when the port is a trunk port.
- **Off** indicates the port is reserved for IP phones, for conferencing, or is unconfigured.

The following describes the switch port LED behavior and meaning:

- **Telephone Port LEDs (Green)**

— **When on steady**, the port is configured as a telephone port and the telephone is idle.

— **When flashing with ring cadence**, the telephone is ringing.

— **When flashing slowly**, the telephone is off hook.

— **When flashing fast**, the port is in use (call in progress).

- **Trunk Port LED (Yellow)**

— **When on steady**, the port is configured as a trunk port and the trunk is idle.

— **When flashing slowly**, the trunk is off hook.

— **When flashing fast,** the trunk is in use (call in progress).

- **Port LED Alternating Green/Yellow:** The port is out of service. The LED periodically alternates green/yellow or yellow/green. The LED color between alternating colors indicates the port type: green for phone and yellow for trunk.
- **Off (IP phone):** When the LED is off, the port is reserved for IP phones, for conferencing, or is unconfigured.

Network LEDs

The network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:

— **When flashing**, network activity is detected.

— **When on (not flashing)**, heavy network activity is detected.

— **When off**, the switch has no power.

This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- **100M**

— **When green**, the switch is connected to a 100BaseT network.

— **When off**, the switch is connected to a 10BaseT network.

C.4.3.3 ShoreTel Voice Switch 60 Connectors

ShoreTel Voice Switch 60 voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-11 connector for connecting an analog phone (extension 9)

- 1 RJ-21X male connector for mass termination of the telephone/trunk ports
  - — **Power Failure Transfer Unit:** Trunk Port 8 to Extension Port 9
  - — **Backup Operator:** Extension Port 9

ShoreTel Voice Switch 60 RJ-21X Telephone and Trunk Connector

Table C-12 lists the RJ-21X Ring and Tip pin numbers for the SG 60.

Table C-12 ShoreTel Voice Switch 60 RJ-21X Telephone and Trunk Connector Pins

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
1	Trunk, DID, Extension	1	Blue/White	26	White/Blue
2	Trunk, DID, Extension	2	Orange/White	27	White/Orange
3	Trunk, DID, Extension	3	Green/White	28	White/Green
4	Trunk, DID, Extension	4	Brown/White	29	White/Brown
5	Trunk, DID, Extension	5	Slate/White	30	White/Slate
6	Trunk, DID, Extension	6	Blue/Red	31	Red/Blue
7	Trunk, DID, Extension	7	Orange/Red	32	Red/Orange
8	Trunk, DID, Extension	8	Green/Red	33	Red/Green
9	Extension	9	Brown/Red	34	Red/Brown
10	Extension	10	Slate/Red	35	Red/Slate
11	Extension	11	Blue/Black	36	Black/Blue
12	Extension	12	Orange/Black	37	Black/Orange
–		13	Green/Black	38	Black/Green
–		14	Brown/Black	39	Black/Brown
–		15	Slate/Black	40	Black/Slate
–		16	Blue/Yellow	41	Yellow/Blue
–		17	Orange/Yellow	42	Yellow/Orange
–		18	Green/Yellow	43	Yellow/Green
–		19	Brown/Yellow	44	Yellow/Brown
–		20	Slate/Yellow	45	Yellow/Slate
–		21	Blue/Violet	46	Violet/Blue

Table C-12 ShoreTel Voice Switch 60 RJ-21X Telephone and Trunk Connector Pins (Continued)

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
–		22	Orange/Violet	47	Violet/Orange
–		23	Green/Violet	48	Violet/Green
–		24	Brown/Violet	49	Violet/Brown
–		25	Slate/Violet	50	Violet/Slate

C.4.4 ShoreTel Voice Switch 40

The following sections describe ShoreTel Voice Switch 40 resource capacity, LED behavior, and connectors. The ShoreTel Voice Switch 40 is not supported in installations outside the U.S. and Canada. Figure C-18 displays the ShoreTel Voice Switch 40 front plate.

The ShoreTel Voice Switch 40 is often referred to as the ShoreTel 40/8 (SG 40/8).

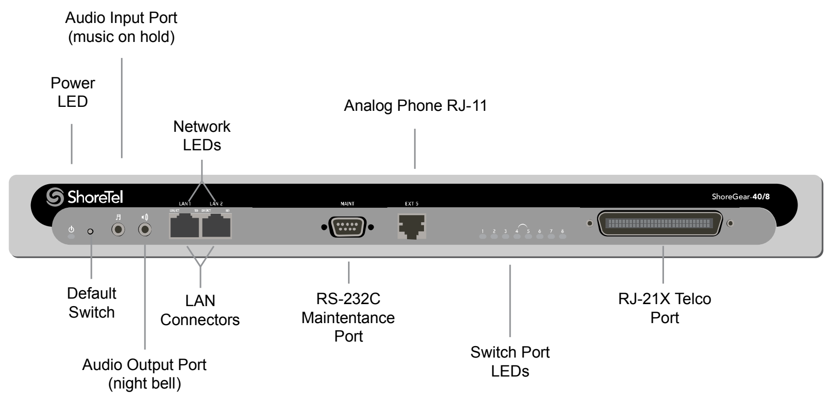


Figure C-18 ShoreTel Voice Switch 40 Front Plate

C.4.4.1 Switch Capacity

- **Analog Circuit Resources**
  - — **Ports 1-2**: Two Loop Start Trunks, DID Trunks, or Extensions
  - — **Ports 3-4**: Two Loop Start Trunks.
  - — **Ports 5-8**: Four Extensions
  - — **Power Failure Transfer Unit:** Trunk Port 4 to Extension Port 5
- **Make Me Conference Resource**: eight ports
  - — **Ports 1-8**
- **Maximum IP Phone Resources**: 40 devices
  - — **Analog Port Reallocation:** 40

C.4.4.2 LED Descriptions

Power LED

The ShoreTel Voice Switch 40 has one power LED, which indicates the following:

- **On**: The switch is operating normally.
- **Off**: The switch has no power.
- **Flashing**:
  - **2 flashes—**The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
  - **3 flashes—**Booting via FTP. Flash memory might be corrupted. Go to the Quick Look page to ensure that the system is running properly.
  - **4 flashes—**The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.
  - **5 flashes—**The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreTel server, and set the boot file to `/tsa/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreTel server's IP address, and set option 67 to `/tsa/vxworks`.
  - **6 flashes—**Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Switch Port LEDs

The ShoreTel Voice Switch 40 has 8 telephone/trunk port LEDs. The color of the LED indicates the port function:

- **Green** when the port is a telephone port.
- **Yellow** when the port is a trunk port.
- **Off** indicates the port is reserved for IP phones, for conferencing, or is unconfigured.

The following describes the switch port LED behavior and meaning.

- **Telephone Port LEDs (Green)**
  - **When on steady**, the port is configured as a telephone port and the telephone is idle.
  - **When flashing with ring cadence**, the telephone is ringing.

— **When flashing slowly**, the telephone is off hook.

— **When flashing fast**, the port is in use (call in progress).

- **Trunk Port LED (Yellow)**

— **When on steady**, the port is configured as a trunk port and the trunk is idle.

— **When flashing slowly**, the trunk is off hook.

— **When flashing fast**, the trunk is in use (call in progress).

- **Port LED Alternating Green/Yellow:** The port is out of service. The LED periodically alternates green/yellow or yellow/green. The color of the LED between alternating colors indicates the port type: green for phone and yellow for trunk.
- **Off (IP phone):** When the LED is off, the port is reserved for IP phones, for conferencing, or is unconfigured.

Network LEDs

The network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:

— **When flashing**, network activity is detected.

— **When on (not flashing)**, heavy network activity is detected.

— **When off**, network activity is not detected.

This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- **100M**

— **When green**, the switch is connected to a 100BaseT network.

— **When off**, the switch is connected to a 10BaseT network.

C.4.4.3 ShoreTel Voice Switch 40 Connectors

The ShoreTel Voice Switch 40 voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-11 connector for connecting an analog phone (extension 9)

- 1 RJ-21X male connector for mass termination of the telephone/trunk ports
  - — **Power Failure Transfer Unit:** Trunk Port 4 to Extension Port 5
  - — **Backup Operator:** Extension Port 5

ShoreTel Voice Switch 40 RJ-21X Telephone and Trunk Connector

Table C-13 lists the RJ-21X Ring and Tip pin numbers for the SG 40I.

Table C-13 ShoreTel Voice Switch 40 RJ-21X Telephone and Trunk Connector Pins

		Ring		Tip	
Port	**Type**	**Pin #**	**Cable Color**	**Pin #**	**Cable Color**
–		1	Blue/White	26	White/Blue
–		2	Orange/White	27	White/Orange
–		3	Green/White	28	White/Green
–		4	Brown/White	29	White/Brown
1	Trunk, DID, Extension	5	Slate/White	30	White/Slate
2	Trunk, DID, Extension	6	Blue/Red	31	Red/Blue
3	Trunk	7	Orange/Red	32	Red/Orange
4	Trunk	8	Green/Red	33	Red/Green
5	Extension	9	Brown/Red	34	Red/Brown
6	Extension	10	Slate/Red	35	Red/Slate
7	Extension	11	Blue/Black	36	Black/Blue
8	Extension	12	Orange/Black	37	Black/Orange
–		13	Green/Black	38	Black/Green
–		14	Brown/Black	39	Black/Brown
–		15	Slate/Black	40	Black/Slate
–		16	Blue/Yellow	41	Yellow/Blue
–		17	Orange/Yellow	42	Yellow/Orange
–		18	Green/Yellow	43	Yellow/Green
–		19	Brown/Yellow	44	Yellow/Brown
–		20	Slate/Yellow	45	Yellow/Slate
–		21	Blue/Violet	46	Violet/Blue
–		22	Orange/Violet	47	Violet/Orange
–		23	Green/Violet	48	Violet/Green
–		24	Brown/Violet	49	Violet/Brown
–		25	Slate/Violet	50	Violet/Slate

C.4.5 ShoreTel Voice Switch T1 and ShoreTel Voice Switch E1

The following sections describe ShoreTel Voice Switch T1 and ShoreTel Voice Switch E1 resource capacity, LED behavior, and connectors. The ShoreTel Voice Switch T1 is not supported in installations outside the U.S. and Canada. Figure C-19 displays the ShoreTel Voice Switch T1 front plate. The ShoreTel Voice Switch E1 front plate is identical to the ShoreTel Voice Switch T1 except for the E1 labeling.

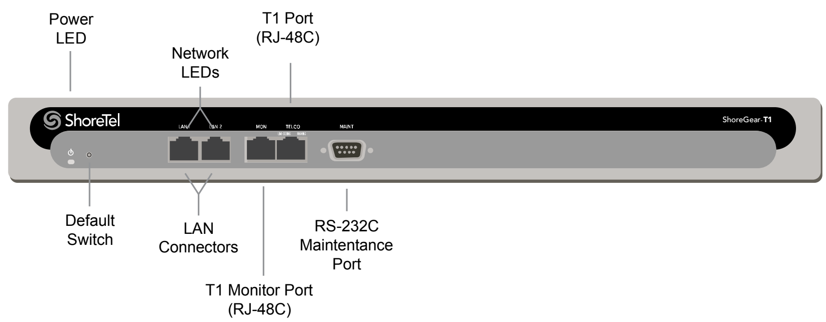


Figure C-19 ShoreTel Voice Switch T1 Front Plate

The ShoreTel Voice Switch T1 provides higher-density trunking to the central office using CAS or PRI signaling. It can also be used as a gateway to legacy PBX systems.

A ShoreTel Voice Switch E1 provides higher-density trunking to the central office using PRI signaling. It can also be used as a gateway to legacy PBX systems.

C.4.5.1 Switch Capacity

- **Digital Circuit Resources**
  - **ShoreTel Voice Switch T1 – One T1 circuit, 24 channels per circuit**: 24 channels maximum
  - **ShoreTel Voice Switch E1 – One E1 circuit, 30 channels per circuit**: 30 channels maximum
- **Make Me Conference Resources:** None
- **Maximum IP Phone Resources:** None

C.4.5.2 LED Descriptions

Power LED

The ShoreTel Voice Switch T1 and ShoreTel Voice Switch E1 voice switches have one power LED, which indicates the following:

- **On**: The switch is operating normally.
- **Off**: The switch has no power.
- **Flashing**:
  - **2 flashes—**The switch failed its internal self-test. Hardware failed; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
  - **3 flashes—**Booting via FTP. Flash memory may be corrupted. Go to the Quick Look page to make sure that the system is running properly.

— **4 flashes—**The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.

— **5 flashes—**The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds.

— **6 flashes—**Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

The network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity. The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:

  — **When flashing**, network activity is detected.

  — **When on (not flashing)**, heavy network activity is detected.

  — **When off**, network activity is not detected.

  This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- **100M**

  — **When green**, the switch is connected to a 100BaseT network.

  — **When off**, the switch is connected to a 10BaseT network.

Monitor and Telco LEDs

The Monitor and Telco LEDs indicate line coding, network framing, and loopback status. These LEDs are color coded—green, yellow, and red. The Monitor and Telco LED descriptions follow.

Telco and Monitor LED alarms and errors are logged as switch events in ShoreTel Director's event log.

- **Line Coding:** This LED indicates line coding status, as follows:

  — **When green**, the line coding signal is good.

  — **When yellow**, bipolar violations (BPV) are being received at one second intervals.

  — **When red**, a loss of signal (LOS) has occurred.

  — **When flashing red**, loopback is active (local or CO).

  — **When off**, the switch has no power.

- **Framing:** This LED indicates network framing status, as follows:
  - **When green,** the T1/E1 signal is in frame; the signal is synchronized.
  - **When yellow**, a yellow alarm has been received from the Central Office.
  - **When flashing yellow**, the frame-bit error rate has exceeded its limits.
  - **When flashing slow yellow**, a small number of frame-bit errors (>1 per million) have occurred; this state will take up to 10 minutes to clear.
  - **When flashing fast yellow**, a series of frame-bit errors (>1 per 1000) have occurred.
  - **When red**, the T1/E1 signal is out-of-frame (OOF). The received signal cannot be framed to the Extended Superframe (ESF) or D4 format.
  - **When flashing red**, loopback is active (local or CO).
  - **When off**, the switch has no power.

ShoreTel Voice Switch T1 and ShoreTel Voice Switch E1 Connectors

The ShoreTel Voice Switch T1 and ShoreTel Voice Switch E1 voice switches contain the following components:

- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-45 T1 telco port
- 1 RJ-45 T1 monitor port for connecting test equipment

APPENDIX D

Centralized Dial Number (DN)

ShoreTel supports Centralized Dial Numbers (DN). Centralized DN guarantees the data integrity for DN references within the system. When administrators delete a particular DN, Centralized DN checks all the references to that DN across the system. Depending upon the significance of references to the DN to be deleted, the system either allows the deletion by removing all the DN references or prevents the deletion by prompting administrator with a message indicating the referenced DN and that it cannot be deleted.

To delete non-significant references together with the DN eliminates the unnecessary pop-up messages when administrators delete a unwanted DN. It makes smoother and easier DN management for administrator. Table D-1 provide information about the centralized dial number table as it relates to the ShoreTel system.

Table D-1 Centralized Dial Numbers (DN) Table

Deleting an Extension	System Behavior
Hunt Group	---
• Backup Extension	**deletion not available**
• CF destination - call stack full	**delete silently**
• CF destination - no answer	**delete silently**
Members	**delete silently**
Escalation Profile	---
• Automatic Message Forwarding	**delete silently**
• Escalation Step (Notification Number)	**delete silently**
Find Me	---
• Primary destination	**delete silently**
• Backup destination	**delete silently**
Users	---
• Mailbox for Recorded Calls	**delete silently**
• Delayed Ringdown	**delete silently**
Programmable Button	---
• Toolbars	**delete silently**

Table D-1 Centralized Dial Numbers (DN) Table (Continued)

Deleting an Extension	System Behavior
User Call Handling Modes	--- -
• Always Destination:	set to DN Type =4
• Busy Destination	set to DN Type =4
• No Answer Destination	set to DN Type =4
• Personal Assistant	delete silently
Extension List	--- -
• User Ext List	delete silently

User Call Handling Modes Defaults	--- -
• Always Destination:	set to DN Type =4
• Busy Destination	set to DN Type =4
• No Answer Destination	set to DN Type =4
• Personal Assistant	delete silently
• Call Handling Modes Delegation	delete silently
Bridge Call Appearance (BCA)	--- -
• Backup Extension	delete silently
• Call Stack Full	delete silently
• No Answer	delete silently
System Dist Lists	delete silently
WorkGroup	delete silently
• Backup Extension	set to DN Type =4
• Call Handling Modes Work Group Assistant	delete silently
• Work Group Assistant	delete silently
• Members (Work Group Agents)	manual deletion required
• Queue Handling Steps (Operation)	manual deletion required
• Over Flow DN (Queue Handling Steps	manual deletion required
Route Point	--- -
• Call Handling Modes	set to DN Type =4
• Assistant	delete silently
AA Menu	--- -
• Extension in Steps	manual deletion required
• Group Paging	delete silently
• Pickup Group	delete silently

Table D-1 Centralized Dial Numbers (DN) Table (Continued)

Deleting an Extension	System Behavior
AMIS	--- -
• Delivery Number	no impact
• Callback Number	no impact
CoST/ClassofService	--- -
• Directed Pagin	delete silently
• Barge In	delete silently
• Record Other's Calls	delete silently
• Silent Monitor	delete silently

Glossary

802.1x A component of IEEE 802.1 group of networking protocols that addresses port-based Network Access Control. It provides an authentication mechanism for devices attaching to a LAN port to either establish a point-to-point connection or prevent the device from accessing the port if authentication fails. 802.1x is based on the Extensible Authentication Protocol (EAP).

ACD *see Automatic Call Distributor*

Active Directory (AD) *Active Directory* is a Microsoft technology that provides various network services, including LDAP-style directory services and authentication, for Windows environments. Administrators to assign policies, deploy software, and apply critical updates to an organization from a single Active Directory database.

Active Call Handling Mode The active call handling mode is the CHM that is designated to specify the current method of handling a user's inbound calls. One mode is always active for each user. The active CHM is selected manually by the user or changes automatically based on the user's schedule.

AD *see Active Directory*

Administrator, System The IT professional responsible for installing, configuring, and maintaining a ShoreTel system.

Advanced Encryption Standard (AES) *Advanced Encryption Standard* is a United States Encryption standard defined in Federal Information Processing Standard (FIPS) 192, published in Nov. 2001. AES is a symmetric block cipher which can process fixed 128-bit data blocks, using 128, 192 or 256-bit keys.

AES *see Advanced Encryption Standard*

All Trunks Busy The condition where a user is cannot complete a call because no trunks are available. The system denotes an all trunks busy condition with a fast-busy tone.

Announced Find Me Announced Find Me is an option where the identity of an inbound caller is provided to the Find Me call recipient. Inbound callers may be required to record their spoken name before the call is connected.

Automatic Call Distributor (ACD) An Automatic Call Distributor is a system that distributes inbound calls to members, or agents, of the system.

Auto Find Me Auto Find Me is a Find Me option where a call recipient can program the system immediately forward inbound call without requiring the caller to press "1".

Automated Attendant (Auto-Attendant) A program that answers and handles inbound calls without human intervention. Auto attendants typically provide menu-driven options through which callers can obtain information, perform tasks, or connect to a requested extension.

Automated Call Handling A method of managing inbound calls that relies on Call Handling Modes to specify a destination or resolution for calls on the basis of the user's schedule or status.

Available DID numbers DID numbers within a range that are not assigned to a user or entity within the context of that range.

DID number availability within a range does not consider DNIS assignments. Numbers assigned as a DNIS number are still enumerated as ***available*** within a DID range; attempts to assign these DID numbers will be unsuccessful.

Barge In Barge In is a ShoreTel function that allows a user to enter another user's active call. The initiator can listen and speak to all other call participants. A Telephony Class of Service sets Barge In rights.

BCA *see* ***Bridged Call Appearance***

BOOTP (or Boot Protocol or Bootstrap Protocol) A network protocol used by network clients to obtain an IP address from a configuration server. BOOTP is typically used when a computer or system is starting up.

Bounced Call A bounced calls is a parked call that is returned to the extension that parked the call because is was not picked up within a specified period.

Bridged Call Appearance (BCA) A Bridged Call Appearance is an extension that is shared among multiple users. A BCA is characterized by an extension number and its call stack depth. The extension number defines the method of contacting BCA users. The call stack depth specifies the number of call that can simultaneously reside on the BCA.

Bridged Call Appearance Monitor The Bridged Call Appearance Monitor is a ShoreTel Communicator window that displays all Bridged Call Appearances for which the user's extension can access through all devices assigned to the user. The Bridged Call Appearance Monitor is only available through Operator ShoreTel Communicator.

Call Appearance A VoIP data structure that supports one voice call session.

Call Detail Record (CDR) A Call Detail Record (CDR) is a data record containing statistics of a call that passed through a pbx or telephone exchange.

Information provided by a CDR includes the number of the originating party, the number of the recipient, the time that the call started, and the duration of the call. ShortTel CDRs contain additional information.

Call Handling A predetermined method of servicing inbound calls. ShoreTel supports Automated Call Handling and Personalized Call Handling.

Call Handling Mode (CHM) A call handling mode is a ShoreTel variable that defines a method of handling specified inbound calls. Each user is assigned five call handling modes, one of which is active and determines the user's current call handling method. *See Active Call Handling Mode*.

The five call handling modes assigned to each user are named Standard, In a Meeting, Extended Absence, Out of Office, and Custom.

Call Handling Rule A call handling rule is the base unit of a Call Routing Plan that consists of a condition and action. When a call handling rule is active and the condition is satisfied, the action specifies method by which the ShoreTel system handles the user's call.

Call History A set of call records for an end user. Each call history record provides information about one call including type, direction, time, duration, disposition, and identity of the other party. Users view their call history through ShoreTel Communicator or Mobile ShoreTel Communicator.

ShoreTel Communicator (client application) ShoreTel Communicator is the ShoreTel client application that manages a user's calls, voice mail, and personal system settings through a graphical user interface.

ShoreTel Communicator (switch) ShoreTel Communicator is a ShoreTel Voice Switch module that handles MGCP information from the IP phones to which the switch is assigned.

This hardware module is not directly related to the ShoreTel Communicator client application.

Call Notification A set of features that inform the user of the arrival of a new call, such as ringing the telephone or playing a sound on the workstation speakers.

Call Routing A method of delivering calls to destinations based on a situation or system status. Call routing can also refer to the automatic delivery of an incoming call to a particular extension, such as in DID or dedicated CO lines.

Call Routing Plan The Call Routing Plan is the Personal Call Handling component that contains of rules that specifies a user's call handling method. The plan prioritizes a list of enabled Call Handling Rules.

Call Stack The ShoreTel extension component that manages an entity's call appearances. The size defines the maximum number of calls – including active and held calls – that an extension can handle simultaneously.

Call Waiting Usually for single-line telephones, a feature that lets a second call arrive to the line by delivering a call-waiting tone to the user and a ring-back to the caller.

Call Waiting Tone The tone that is presented to a user with call waiting when a second call arrives.

CallerID A technique for transmitting the calling party's telephone number and (optionally) name to equipment enabled to handle this feature; also called CLI in Europe.

Caller's Emergency Service Identification (CESID) The extension that a switch sends to a Public Safety Answering Point (PSAP). A CESID identifies the location of callers who require emergency 911 services.

CDR *see Call Detail Record*

CDR Database A CDR database contains all CDRs generated by a ShoreTel system over a specified period of time. The CDR database is maintained on the Main Server.

Central Office (CO) The building where the telephone company's telephone switching equipment that services the local area is located.

Centrex A name for advanced telephone services provided by the local telephone company. It usually requires a connection to a special telephone system but provides services such as voice mail and call forwarding.

CESID *see* ***Caller's Emergency Service Identification.***

CHM *See* ***Call Handling Mode***.

Class of Service (CoS or COS) A class of service grants permission to a specified set of features and privileges. Users assigned to a class of service can access the specified features and privileges. ShoreTel defines three types of service classes: telephony features, call permissions, and voice mail permissions.

CO *See* ***Central Office***.

CO Line *See* ***Trunk***.

Codec A codec is a device or program that encodes and decodes a digital data stream or signal. Codecs encode a data stream or signal for transmission, storage or encryption and decode it for viewing or editing. Codecs are typically used in video conferencing and streaming media applications.

Conference Three or more parties joined together in a single call, such that each party can hear and be heard by the others.

Coordinate Location Based Services data point that identifies a specific location in terms of longitude, latitude, and elevation. *See Location Based Services*.

CoS or COS *See **Class of Service***.

Current Call Handling Mode *see Active Call Handling Mode*.

DBImport DBImport is a ShoreTel utility that uses the contents of a CSV file to update the ShoreTel user database. DBImport.exe adds, deletes, and modifies user account records based on the CSV file contents.

DHCP *See **Dynamic Host Configuration Protocol***.

Dial Number (DN) Dial Number – the number dialed by a user.

Dialed Number Identification Service (DNIS) A service from Telcos that lets an enterprise determine which telephone number was dialed by a customer.

DID *see Direct Inward Dialing*

DID Range A DID range is a list of consecutive DID numbers assigned to a Trunk Group. Ranges assigned to a trunk group cannot overlap. A DID number can be assigned to multiple trunk groups.

Differentiated Services Differentiated Services is a computer networking architecture that specifies a method for managing network traffic and providing Quality of Service (QoS) guarantees on IP networks.

Differentiated Services can provide low-latency, guaranteed service for critical network traffic (voice or video) while providing simple best-effort traffic guarantees to less critical services (web traffic or file transfers).

Differentiated Services Code Point (DSCP) Differentiated Services Code Point is a 6-bit value, contained in the 8-bit DiffServ/ToS field of the IP packet header, that indicates the data traffic Per-Hop Behavior (PHB).

DN *see dial number.*

DNIS *see Dialed Number Identification Service*.

DSCP *see Differentiated Services Code Point*.

DTMF *See **Dual-Tone Multi-Frequency**.*

Dynamic Host Configuration Protocol (DHCP) A protocol for downloading network information (such as IP addresses) to client workstations.

Dual-Tone Multi-Frequency (DTMF) A signalling method used by telephones to send information to an exchange. Pressing a key on the phone's keypad sends two simultaneous tones to the exchange, which are then decoded to determine the pressed key. Also known as Touch Tone.

Explicit Authentication Explicit Authentication is an LDAP process where an application prompts users for credentials when they log into a network, then verifies those credentials through a call to the LDAP server.

Extension Assignment Extension Assignment is a ShoreTel user feature that forwards calls to a designated device – either a system IP phone device or a remote telephone connected to the PSTN. Extension Assignment requires administrative authorization.

Extension Monitor Extension Monitor is a ShoreTel function that provides a user access to another user's extension. In addition to viewing extension activity, the monitoring user can answer calls and perform other actions on calls at that extension. Extension monitoring is facilitated through ShoreTel Communicator and IP Phone Programmable buttons.

Failback Failback is the process of restoring a server, system, component, or network to the original, pre-failover state. *See failover.*

Failover Failover is the action of automatically transferring operations to a redundant or standby server, system, component, or network to continue normal operation after the base device fails. The failover process initiates without human intervention and typically without warning.

Find Me Find Me is a call handling feature that allows callers who are routed to a user's voice mailbox to contact the user at alternate devices by pressing "1" while listening to the voice mail greeting. See *Auto Find Me* and *Announced Find Me*.

Foreign Exchange Office (FXO) An FXO interface connects to the public switched telephone network (PSTN) central office and is the interface offered on a standard telephone. An FXO interface is used for trunks, tie lines, or connections to a PSTN CO or PBX that does not support E&M signaling (when local telecommunications authority permits).

Foreign Exchange Station (FXS) An FXS interface supplies ring, voltage and dial tone for basic telephone equipment, keysets, and PBXs. The FXO interface is useful for off-premises station applications.

Frequency Shift Key (FSK) A modulation technique used with low-speed modems; also used with CallerID and message-waiting lamp indicators.

FSK *See* ***Frequency Shift Key****.*

FXO *See* ***Foreign Exchange Office***.

FXS *See* ***Foreign Exchange Station***.

Global Positioning System The Global Positioning System (GPS) uses a constellation of 24 and 32 satellites that transmit microwave signals to GPS receivers, which they use to determine their current location, the time, and their velocity. GPS usually requires line of sight access between the transmitter and the device.

GPS *see Global Positioning System*

Hold (on Hold) A active call whose conversation is suspended. Calls on hold remain on the user's.

IM *see Instant Messaging*

Instant Messaging (IM) Instant Messaging is the real-time transmission of text between at least two parties.

Internet Telephony Service Provider (ITSP) An *Internet Telephony Service Provider* are vendors that offer Internet data services for making telephone calls.

IP Phone Configuration Switch In each ShoreTel installation, one switch is responsible for assigning, to each IP phone, a switch that performs ShoreTel Communicator activities.

IP Phone Failover IP Phone Failover is a ShoreTel feature that continues a voice call after the ShoreTel IP Phone loses communication with the its ShoreTel Communicator switch.

When IP Phone Failover is implemented, ShoreTel IP Phones are configured to send a signal to their ShoreTel Voice Switch every four minutes and to expect an acknowledgment from the switch. When a phone does not receive the return signal, it connects to another switch located at the same site.

IP Phone Keep Alive A signal that voice switches send to the IP phones for which they provide ShoreTel Communicator services. The system considers phones that acknowledge the signal are The signal is also referred to as a heartbeat.

ITSP *see Internet Telephony Service Provider*

LBS *see Location based service.*

LDAP *see Lightweight Directory Access Protocol.*

LDAPExport LDAPExport is a ShoreTel utility that exports AD directory information to a CSV file, whose contents can be imported to a ShoreTel user database. LDAPExport was introduced in ShoreTel to support AD Integration.

Lightweight Directory Access Protocol (LDAP) *Lightweight Directory Access Protocol* is an application protocol for querying and modifying directory services running over TCP/IP.

Line *See* ***Trunk****.*

LLDP-MED An enhancement to the basic LLDP protocol that addresses the discovery of endpoints by networks supporting LLDP.

Location based service Location based service (LBS) is a component feature that receives data identifying the position of the device in terms of latitude, longitude, and altitude. Data reception methods include GPS access and through Mobile Cell device service transmitters.

MCM location based services does not use altitude to specify the location of the device.

Loop Start One of the mechanisms used to signal the telephone system that the calling party wants to make a call. Loop start is a completion of the circuit using a set load between the two wires (tip and ring).

Message Notification A set of features that inform the user that a new message has arrived in his or her voice mailbox, such as lighting the call-waiting lamp, paging the user, or dialing a telephone number.

MOH *See* ***Music on Hold****.*

Monitor Extension Extension monitoring is a ShoreTel feature that permits a user to view another user's extension status and answer inbound or held calls to that extension.

Music on Hold (MOH) Background music heard when callers are put on hold, letting them know they are still connected. Most telephone systems have the ability to connect to any sound-producing device—for example, a radio, a cassette, or a CD player.

MySQL MySQL is a relational database management system. that runs as a server providing multi-user access to a number of databases. MySQL is owned and sponsored by a single for-profit firm, the Swedish company MySQL AB, which is a subsidiary of Sun Microsystems. The project's source code is available under terms of the GNU General Public License, as well as under a variety of proprietary agreements.

MySQL is officially pronounced *My S Q L*, not *My sequel*. This adheres to the official ANSI pronunciation.

ODBC The *Open Database Connectivity* specification provides a standard software API method for using database management systems (DBMS). The designers of ODBC aimed to make it independent of programming languages, database systems, and operating systems.

ODBC Connectors Drivers that provide connectivity to the MySQL server for client programs. There are currently five MySQL Connectors – Connector/ODBC, Connector/NET, Connector/J, Connector/MXJ, Connector/PHP.

Off Hook The operating state of a communications link in which transmissions are enabled either for network signaling, voice communications, or data communications.

The act of seizing the line or channel is referred to as going off hook.

Off System Extensions Off System Extensions are ShoreTel extensions that, when dialed, route calls from the trunk group to which they are associated. A ShoreTel system typically uses Off System Extensions to connect with a legacy PBX system.

On Hook The operating state of a telecommunications link in which transmissions are disabled and then end instrument (phone) presents an open circuit to the link. The link is responsive to ringing signals during on-hook conditions.

The act of releasing the line or channel is referred to as going on hook.

Operator The person who monitors the telephone system and transfers calls to the appropriate extensions.

Parking a Call Parking a call places a call on hold on another extension's.

PBX *See* ***Private Branch Exchange***.

Per-Hop Behavior Per-Hop behaviors define packet forwarding properties associated with a class of traffic. Different PHBs may be defined to offer low-loss, low-latency forwarding properties or best-effort forwarding properties.

Personalized Call Handling Personalized Call Handling is a ShoreTel feature that defines flexible handling methods for inbound calls. Calls are filtered and managed on the basis caller identity, time or date or receipt, end user status, and caller identity.

Physical Extension A common internal extension with an assigned physical port and telephone.

Presence Presence is a feature that identifies and distributes the availability of system users and other personal contacts. ShoreTel defines two types of Presence: IM presence and telephony presence.

Private Branch Exchange (PBX) A term used by telephone companies to indicate equipment that is located on the customer's premises and that can route telephone calls.

Programmable ShoreTel Communicator Toolbar Buttons Programmable Toolbar buttons is a ShoreTel Communicator feature that provides access to ShoreTel call management functions. Each User Interface button is programmed by the administrator to perform a specific function when clicked and contains text that provides status of the function.

Programmable IP Phone Buttons Programmable IP Phone buttons is a ShoreTel IP Phone feature that provides access to ShoreTel call management functions. Each button is programmed by the user or administrator to perform a specific function when pushed and contains an LED indicator that displays status of the function.

PSTN *See **Public Switched Telephone Network**.*

Public Switched Telephone Network (PSTN) Another name for the public telephone network.

QoS A traffic engineering term that refers to resource reservation control by providing different priorities to different applications, users, or data flows to guarantee a performance level for a data flow.

QuickDial Field A QuickDial field is a data entry field requesting contact information for a system user, extension, or external contact. Upon the entry of alphanumeric characters, the field expands vertically to display all possible valid entries, filtered by the input. Users fill the field by selecting one of the options with the cursor.

Real-time Transport Control Protocol (RTCP) *Real-time Transport Control Protocol*, defined in RFC 3550, is a sister protocol of the Real-time Transport Protocol (RTP) that provides out-of-band control information for an RTP flow. It accompanies RTP in the delivery and packaging of multimedia data, but does not transport any data itself. It is used periodically to transmit control packets to participants in a streaming multimedia session, primarily to provide feedback on the quality of service being provided by RTP media stream.

Real-time Transport Protocol (RTP) *Real-time Transport Protocol* defines a standardized packet format for delivering audio and video over the Internet. RTP is described by IETF RFC 3550.

Ring *See **Tip and Ring**.*

Ringback Tone The audible signal given to the caller by the telephone company (or telephone system) to indicate that the remote telephone is ringing.

Ringdown A ringdown circuit consists of predefined devices at the circuit endpoints and is configured to ring a recipient device immediately after an initiating device goes off hook. Ringdown calls are completed without dialing or any other signaling other than the initiating device going off hook.

RTCP *see Real-time Transport Control Protocol.*

RTP *see Real-time Transport Protocol.*

SDP *see Session Description Protocol.*

Secure Real-Time Transport Protocol (SRTP) *Secure Real-Time Transport Protocol* defines a profile for providing encryption, message authentication and integrity, and replay protection to RTP data streams.

Session Description Protocol (SDP) *Session Description Protocol* is a format for describing streaming media initialization parameters. The IETF published a revised specification as an IETF Proposed Standard as RFC 4566 in July 2006. SDP is intended for describing multimedia communication sessions for the purposes of session announcement, session invitation, and other forms of multimedia session initiation. SDP does not provide the content of the media form itself but simply provides a negotiation between two end points to allow them to agree on a media type and format.

Session Initiated Protocol (SIP) *Session Initiation Protocol* is a signalling protocol that is typically used for setting up and tearing down multimedia communication sessions including voice and video calls. SIP also supports video conferencing, streaming multimedia distribution, instant messaging, and presence information.

Silent Monitor Silent Monitor is a ShoreTel function that allows a user to listen to another user's call without being heard by any of the call participants. When a call is monitored, external callers may hear a monitor tone; system users are not signaled. A Telephony Class of Service sets Silent Monitoring rights.

Single Sign On (SSO) *Single Sign On* (SSO) is an LDAP process where applications automatically authenticate LDAP users that are logged into the network domain through their current network credentials. Such users logged into the network are not prompted to re-enter their credentials.

SIP *see Session Initiated Protocol.*

SIP Extensions SIP Extensions are ShoreTel extension provide calling services through SIP devices. By default, ShoreTel extensions support MGCP devices.

SIP Trunk A SIP trunk is an ITSP service that supports business VoIP sessions to endpoints outside of an enterprise network through the connection that accesses the Internet.

Site A site is a ShoreTel data structure that is identified by a geographic location and characterized by transmission capabilities, device extensions, and other parameters required by ShoreTel devices. Switches, servers, users, and other ShoreTel entities are associated with a site.

SRTP *see Secure Real-Time Transport Protocol.*

SSO *see Single Sign On.*

Stutter Tone An intermittent dial tone provided by the telephone system (as opposed to the usual constant dial tone); sometimes used to indicate to the user that there are messages in his or her voice mailbox or that a feature (such as call forwarding) is enabled.

System Database The ShoreTel system database is the data structure that retains system operational information concerning user accounts, system architecture, switches, servers, sites, and status. Changes made through ShoreTel Director are stored in the system database.

T-1 A digital transmission link with a capacity of 1.554 Mbps. A T-1 trunk can normally handle 24 voice conversations, each digitized at 64 Kbps. T-1 lines are used for connecting networks across remote distances.

TAPI *See* ***Telephony Application Programming Interface***.

Telephony Application Programming Interface (TAPI) A telephony software interface included in Microsoft Windows operating system that supports the incorporation of telephony control by other applications.

Tip and Ring Telephony terms for the two wires of an ordinary telephone wire. Tip is the ground side (positive) and Ring is the battery side (negative) of the phone circuit.

ToS An eight-bit field in the IP Packet Header, previously referred to as Type of Service, that is used to accommodate applications that require real-time data streaming, as specified by RFC 3168. The ToS fields contains a six-bit Differentiated Services Code Point and a two-bit Explicit Congestion Notification field.

Trunk Sometimes used synonymously with line or CO line. Traditionally, a trunk from the telephone company connects to a PBX only, and not to a telephone, whereas a line from the telephone company connects to a telephone. For documentation purposes, either term can be used when referring to voice connections from the telephone company.

Trunk Hunt Group A term sometimes used to indicate a group of telephone lines configured by the telephone company to rotate incoming calls among all the lines in search of the next available one. In this way, a company can give out one main number, and all calls to that number will hunt for the next available line or trunk.

Unparking a Call Unparking a call is the retrieval of a parked call to the extension that parked the call.

Waypoint A known location specified by coordinates, associated with a coverage area (radius), and identified by a descriptive label.

Whisper Page Whisper Page is a ShoreTel client feature that allows a user to interrupt an active call and speak with an internal user. The other call party and the initiator are not connected – they cannot hear or speak with each other. A Telephony Class of Service sets Whisper Page rights.

Workgroup Workgroups is a ShoreTel entity that performs ACD functions for inbound calls. Calls are routed to a workgroup through an extension which, in turn, routes them to workgroup agents. Each workgroup is assigned an extension, mailbox, and other parameter settings.

Workstation A personal computer (PC) or similar computer.

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