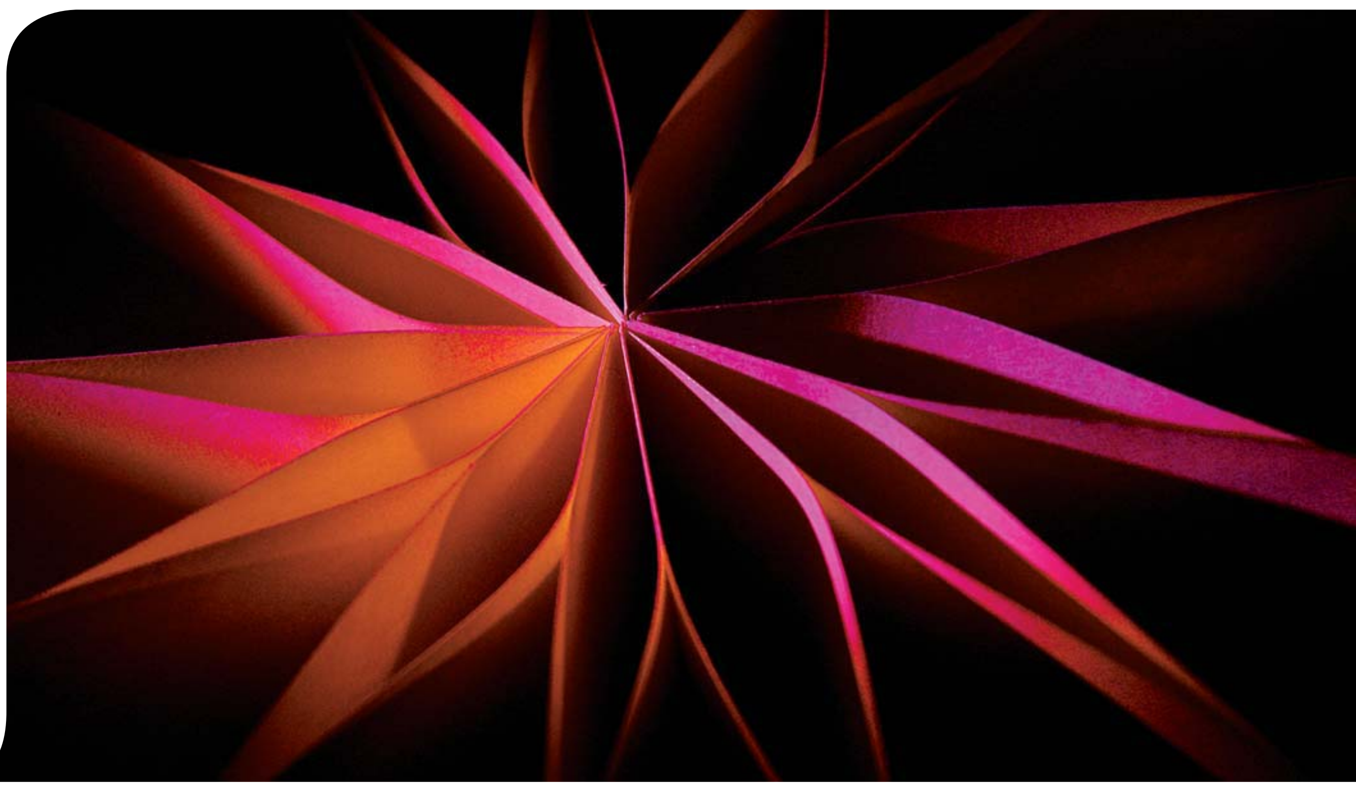
ShoreTel University

ShoreTel 11.1

ShoreTel Planning and Installation Guide



USER GUIDES

RELEASE 11.1

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Table of Contents

Preface

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

Objectives

This document provides planning and installation information for the ShoreTel system and components.

Audience

This guide is written for the person who plans, installs, administers, and maintains the ShoreTel system. This individual should be knowledgeable about data networking and telephony to use this guide effShoreTel 11ectively.

Organization

This document is generally organized into major tasks, presented in the order in which they should be completed.

Documentation Overview

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

System Documentation

The ***ShoreTel Planning and Installation Guide*** (this guide) can be found in the documentation folder on the ShoreWare DVD and can also be accessed from ShoreWare Director.

This 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.

Software Documentation

The ***ShoreTel Administration Guide*** provides detailed reference information (both task-based and screen-by-screen) on how to administer and maintain the ShoreTel system using ShoreWare Director. If you are installing one or more ShoreTel Conference Bridges, refer to the *ShoreTel Converged Conference Solution Administration Guide* for complete installation and configuration information. Both guides can be found in the documentation folder on the DVD.

The following release notes can be found in the documentation folder on the associated DVD and may also be accessed from ShoreWare Director:

- *ShoreTel Release Notes* provide information about new releases, new features, installation, and upgrading for the ShoreWare server.

Hardware Documentation

The following installation documents are packaged with the associated ShoreGear voice switch, conferencing bridge, or ShorePhone IP phone.

- *Quick Install Guides for each ShoreGear switch*
- *ShoreTel Conference Bridge Quick Install Guide*
- *Quick Install Guide for each ShorePhone IP phone*

User Documentation

End-user documentation is installed during the ShoreTel Communicator installation. It is available through the **Help > Contents and Index** command within the ShoreTel Communicator application.

- *Analog Phone Quick Reference*, which is available in the telephone user interface
- *IP Phone Quick Reference*, which is available in the telephone user interface

Release Notes

The release notes listed below provide information about new releases and new features as well as installation and upgrade information. They can be found in the documentation folder on the associated DVD and can also be accessed from ShoreWare Director.

- *ShoreWare Server Release Notes*
- *ShoreWare Client Release Notes*

Online Knowledge Base

To access additional information about the current release or to resolve issues with the ShoreTel system, you can use the ShoreTel online knowledge base. This password-protected, online database is accessible to authorized contacts through the ShoreTel web site at www.ShoreTel.com.

Document Conventions

Conventions used in this guide include the following:

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

CHAPTER 1

Getting Started

Congratulations on your purchase of the ShoreTel system!

Highly flexible, your new ShoreTel system is also simple to install, administer, and maintain. You will be able to unify all your locations and voice applications into a single, efficient voice communications network.

Voice communications is a mission-critical application. This planning and installation guide leads you through the installation process to a successful implementation, so that you and your user community can enjoy the benefits of the ShoreTel system.

Each chapter in this guide begins with recommendations that help you make a smooth transition to the ShoreTel system.

If you are planning an international deployment, please see Appendix A, starting on page 263, for the international capabilities of the ShoreTel system.

1.1 Checklist

Review the following topics before proceeding to the next chapter:

	Task	Description
❑	Recommendations	page 15
❑	Assembling the Team	page 16
❑	ShoreCare ControlPoint	page 16

Table 1-1 Getting Started Checklist

1.2 Recommendations

The following recommendations help ensure that your planning and installation of the ShoreTel system proceeds smoothly.

- **Resource planning:** Do not underestimate the amount of resource commitment needed to successfully implement a mission-critical application such as a new voice communications system.
- **Schedule planning:** Likewise, do not underestimate the amount of time needed to successfully implement the system. Plan necessary tasks ahead of time. Many tasks have long lead times (for example, ordering telephone service, preparing cabling, and ordering telephones), and unforeseen problems can arise that must be resolved.
- **Delegation:** Do not try to do everything yourself. Make sure you assign the right resources to the right task.

Communication: Make sure you communicate with the key members of your organization and determine their individual and departmental needs (whether workgroups, operators, assistants, or executives). Make sure they support any decision that affects their respective areas.

Once the system is successfully deployed, you need to establish clear ownership of the voice communications system. Not only will you want to adapt the system to your changing corporate needs, but you also need to account for the interaction between your data network and your voice application. When changes are made to the data network (for example, renumbering your IP addresses or changing your backbone), you need to consider the impact on your voice communications system, and plan accordingly.

1.3 Assembling the Team

To deploy the ShoreTel system successfully, you need to assemble a team. The key members of the team include, but are not limited to:

Project Manager: Someone needs to oversee the entire project to make sure that key decisions are made and communicated to the entire team, deadlines are met, and issues are resolved. This is typically an IT manager.

System Designer: Someone needs to take ownership of the design of the system, including the number of telephones, number of trunks, and desired call flow. This person is also responsible for the day-to-day system administration after the cut-over to the new system. This is typically a member of the IT staff.

IT Manager: You need the full support and cooperation of your IT department, since the ShoreTel system is a new application on your data network, interacting with servers, desktops, the IP address space, switches, routers, and so on.

Cabling Contractor: You may need to hire a cabling contractor to install racks and cabling, as well as to place and test telephones.

Electrical Contractor: You may need to hire an electrician to install new power outlets, and potentially some cooling and ventilation systems.

Service Providers: You should establish a relationship with a telephone service provider for local and long-distance telephone service. You also need to work with a network service provider to provide IP connectivity between multiple locations, if you have multiple sites.

ShoreTel: Depending on what type of installation and support package you purchased, ShoreTel, or a certified ShoreTel partner, may be involved in your implementation.

1.4 ShoreCare ControlPoint

Installation services are built around ShoreCare ControlPoint, an interactive, web-based project management tool that allows you to take complete control of the installation process. ShoreCare ControlPoint provides real-time visibility into each step of the system installation, from initial needs assessment and resource planning to the final step of going live with the new voice system. It also lets you simultaneously manage installations at multiple sites.

This planning and installation guide can be used in conjunction with ShoreCare ControlPoint. ShoreCare ControlPoint provides step-by-step checklists for each phase of installation and cut-over.

Phase 1: Voice Communications System Analysis and Ordering

	Task	Date Completed
❑	Download and modify the Microsoft Project installation schedule included in Resources	
❑	Complete Call Flow Analysis	
❑	Inventory and determine trunk requirements	
❑	Order new trunk lines	
❑	Trunk installation date	
❑	Inventory your existing telephone equipment	
❑	Order new phones and/or headsets	
❑	Review your need for a ShoreTel Conference Bridge	
❑	Order a ShoreTel Conference Bridge	
❑	Review your need for a ShoreTel Contact Center Solution	
❑	Order a ShoreTel Contact Center Solution	
❑	Order ShoreGear voice switches	
❑	ShoreGear shipping date	

Phase 2: Environmental and Infrastructure Analysis and Upgrade

	Task	Date Completed
❑	Participate in the Phase 2 conference call	
❑	Read ShoreTel's power requirements	
❑	Order power upgrades (as necessary)	
❑	Scheduled power upgrade completion date	
❑	Read ShoreTel's racking requirements	
❑	Racking installation date (if racking is ordered)	
❑	Read ShoreTel's ventilation requirements	
❑	Ventilation system upgrade completion date (if ordered)	
❑	Read ShoreTel's recommendations for Uninterruptable Power Source (UPS)	
❑	UPS installation date (if ordered)	
❑	Read ShoreTel's cabling requirements	
❑	Cabling installation date (if ordered)	
❑	Determine your overhead paging needs	
❑	Source your Music on Hold needs	
❑	Read ShoreTel's LAN requirements	
❑	Attach LAN topology map	
❑	LAN installation date (if ordered)	
❑	Read ShoreTel's WAN requirements	
❑	Attach WAN topology map	
❑	WAN upgrade installation date (if ordered)	
❑	Read ShoreTel's server requirements	
❑	Order your server for the ShoreTel System	
❑	Server installation date	
❑	Read ShoreTel's desktop requirements	
❑	Desktop software upgrade installation date (if required or ordered)	
❑	ShoreGear scheduled installation date	

Phase 3: Resource Scheduling and Tracking

	Task	Date Completed
❑	Participate in the Phase 3 conference call	
❑	Verify Telco order is on schedule	
❑	Verify phone order is on schedule	
❑	Verify power order is on schedule	
❑	Verify racking order is on schedule	
❑	Verify ventilation order is on schedule	
❑	Verify Uninterruptable Power Source (UPS) order is on schedule	
❑	Verify cabling order is on schedule	
❑	Verify LAN upgrade order is on schedule	
❑	Verify WAN upgrade order is on schedule	
❑	Verify desktop upgrade order is on schedule	
❑	Verify ShoreGear order is on schedule	
❑	Read ShoreTel’s descriptions of the different ShoreTel Communicator applications	
❑	Schedule your System Administration training with ShoreTel	
❑	Order new business cards and business stationary if your phone numbers are changing	
❑	Verify that you have obtain all licenses and license keys for your planned installation.	

Phase 4: System Load and Configuration

	Task	Date Completed
❑	Participate in the Phase 4 conference call	
❑	Verify receipt of ShoreGear equipment	
❑	Reserve IP addresses for your network	
❑	Configure server with the appropriate server operating system	
❑	Load the ShoreGear software	
❑	Enter the database configuration for ShoreGear	
❑	Confirm your ShoreTel System installation and cut-over dates	
❑	Confirm installation and cut-over coverage	
❑	Verify racking is complete	
❑	Verify power is in compliance	
❑	Verify UPS is installed	
❑	Verify cabling is complete	
❑	Verify ventilation upgrade is complete	
❑	Verify new phones and headsets have been delivered	
❑	Verify your System Administrators have been trained	
❑	Schedule training for your Operators and Workgroup(s)	

Phase 5: Installation Readiness Review

	Task	Date Completed
❑	Participate in the Phase 5 conference call	
❑	Upgrade desktops, if necessary, and ensure readiness for Client software installation	
❑	Notify users of the ShoreTel system implementation	
❑	Verify telephone trunk lines are installed and tested	
❑	Verify conference bridge is installed	
❑	Configure on-hour and off-hour schedules for Auto-Attendant menus and Workgroups	
❑	Configure your Workgroups	
❑	Configure your Auto-Attendant menus	
❑	Script and record all Auto-Attendant and department voice mail greetings	

Phase 6: Cut-Over

	Task	Date Completed
❑	Participate in the Phase 6 conference call	
❑	Complete your Cutover Review Checklist	
❑	Send web-based training modules to End Users	
❑	Send TUI guides to End Users	
❑	Verify that Operators are trained	
❑	Verify that Workgroups are trained	
❑	Verify that all phones have been placed and extensions tested	
❑	Verify that existing trunk lines have been swapped and tested	
❑	Verify that End Users have been sent the ShoreTel Client notification	
❑	Cut-over to the ShoreTel System	
❑	Complete your Post Cut-over Survey	
❑	Review ShoreTel Web Center to understand the available ShoreTel Support resources	

CHAPTER 2

System Overview

This chapter presents an overview of the ShoreTel system, including a description of the system capacity, to guide you in planning your solution.

2.1 Checklist

Review the following topics before proceeding to the next chapter:

	Task	Description
❑	ShoreTel Distributed IP Voice Architecture	page 23
❑	Distributed Call Control	page 24
❑	Distributed Applications Platform	page 25
❑	Single System Management	page 25
❑	Integrated Applications	page 28
❑	Desktop Applications	page 34
❑	Voice Switches	page 35
❑	ShoreTel IP Phones and Devices	page 35
❑	System Capacity	page 41

Table 2-1 System Overview Checklist

2.2 ShoreTel Distributed IP Voice Architecture

The ShoreTel system is a completely distributed voice communication solution with no single point of failure, which is layered on top of your IP network. At the heart of the system is the standards-based Distributed IP Voice Architecture (Figure 2-1), which uniquely distributes call control intelligence to voice switches connected anywhere on the IP network. In addition, the Distributed IP Voice Architecture distributes voice applications, including voice mail systems and automated attendants, to servers across locations, rather than centralizing applications at the network core.

The resulting solution provides a single image system for all locations and all voice applications. Multiple PBXs, voice mail systems, automated attendants, or ACD systems—each with their own dedicated management interface—are phone systems of the past. The ShoreTel system is distributed, the voice applications are bundled, and the management interface is integrated.

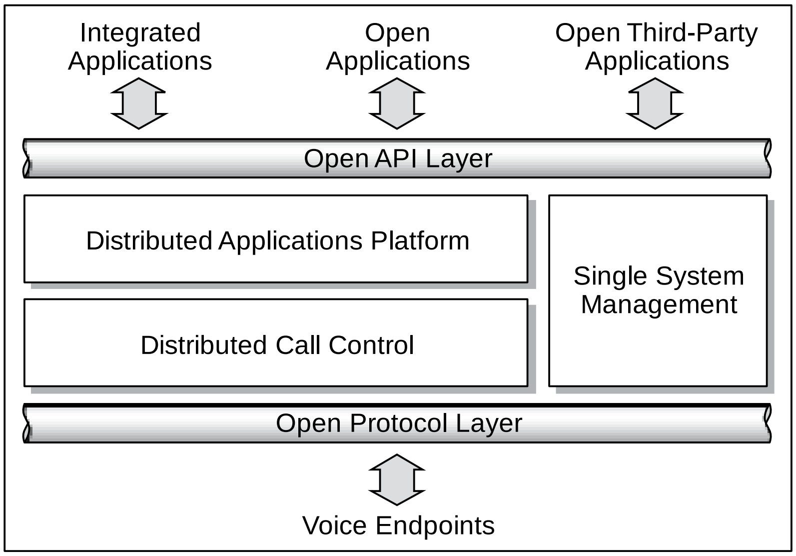

Figure 2-1 The Distributed IP Voice Architecture of the *ShoreTel* System

2.3 Distributed Call Control

The heart of the ShoreTel system is the distributed call control software, which runs on the ShoreGear voice switches on top of VxWorks™ and embedded Linux, a real-time operating system. Each call control element manages the call setup and call teardown, including features such as transfer, conference, forward, call permissions, and call routing. The voice switches communicate on a peer-to-peer basis, eliminating any single point of failure. For instance, if one ShoreGear voice switch goes offline, all other ShoreGear voice switches continue operating. When the voice switch comes back online, it rejoins the voice network with no impact on system operation. There is no server involved with the basic telephony, so the system delivers levels of availability unmatched by even legacy vendors.

2.3.1 Distributed Routing Service

Distributed Routing Service (DRS) allows larger systems to scale beyond 100 switches up to a total of 500 switches (including SoftSwitches). The Distributed Routing Service is optional on systems up to 60 switches, but must be enabled on systems with 60 or more switches.

When the Distributed Routing Service is disabled, ShoreGear switches build an internal routing database from the peer-to-peer communication with other switches. Each ShoreGear switch contains routing information for all endpoints in the system, including information regarding trunk selection for outbound calls. When a user places a call from any extension, each switch can route the call to the correct ShoreGear switch based on its internal routing database.

When the Distributed Routing Service is enabled, ShoreGear switches only exchange routing information with other switches at the same site, rather than exchanging the information with every switch in a multi-site system. Although each ShoreGear switch only maintains routing information within its site, each ShoreWare server also includes an instance of the Distributed Routing Service, which maintains system-wide routing information. When calls are initiated, ShoreGear switches contact the Distributed Routing Service in order to find the ShoreGear switch or switches necessary to complete the call.

In a system with more than one ShoreWare server, the ShoreGear switches may contact an alternate instance of the routing service if the primary instance is not reachable. ShoreWare servers have a hierarchical relationship with the headquarters server at the top of the hierarchy. As you add servers to the system through ShoreWare Director, you define the order of the servers in relation to the headquarters server and the various sites in your system. Initially, the switches try to contact the nearest instance of the Distributed Routing Service in the hierarchy. If that instance of DRS is not reachable, the switch contacts the instance of DRS at the parent server in the hierarchy as a fallback. If both instances of DRS are not reachable, the switch makes a best effort to route the call based on its internal routing tables built from communicating with peer ShoreGear switches at the same site.

2.4 Distributed Applications Platform

The distributed applications platform of the ShoreTel system enables application servers to be distributed across the enterprise yet still behave as a single, cohesive system. This allows you to optimize network performance by locating applications such as voice mail close to users to reduce WAN bandwidth utilization. In addition, by hosting applications, services, and APIs on multiple platforms, the distributed applications platform enables the system to scale as necessary.

A software component called the ShoreWare Telephony Management Service (TMS) runs on the ShoreWare servers and observes all call setup and call teardown activity on the entire voice network. The ShoreWare TMS software then exposes a Telephony Application Programming Interface (TAPI), for call control, and a TAPI Wave interface for media play and record. These open APIs allow value-added applications to be added to the ShoreTel system to provide voice services.

Even though there are multiple application servers, the ShoreTel system is still managed and behaves as a single image system with complete feature transparency between sites.

2.5 Single System Management

The ShoreTel system provides a single system management solution called ShoreWare Director. This browser-based network management tool provides a single management interface for all voice services and applications across all locations. Even though there are multiple servers and switches to support the services and applications, the ShoreTel system provides a single image system across your entire network.

Integrated management enables a change to propagate dynamically across the system each time a modification is made on the ShoreTel system. When you add a new user on the ShoreTel system, the user automatically gets a dialing plan, voice mail, an extension, a mailbox, an Auto-Attendant profile, and an e-mail message to download the desktop software. In addition, the user can be added to an Automated Call Distributor (ACD) group, if needed. You add new users and place them in ACD groups from a single management screen.

The ShoreTel system provides automated software distribution for all components on the system. When you add a new ShoreGear voice switch to the system, it is automatically upgraded to the current software release by the ShoreWare server. When you add a new user on the system, the user receives an e-mail message containing a URL from which desktop call control and unified messaging applications can be download and installed.

For software upgrades, you simply install the new software on the ShoreWare server, and all the ShoreGear voice switches, across all locations, are automatically upgraded to the new release. In addition, users are notified of the new software release and are automatically prompted to upgrade their software, if an upgrade is mandatory.

The ShoreTel management software also provides a complete suite of maintenance tools that enable you to monitor and change the status of components on the system. The system can be configured with event filters that automatically generate an e-mail message if an error occurs on the system.

2.5.1 Multi-level Management

The ShoreTel system provides in-depth access levels to ShoreWare 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 ShoreWare Director can be assigned a level of permission tailored for his needs.

2.6 System Reliability

The ShoreTel system provides a number of features and options that ensure system reliability, including:

- Distributed Switch Control
- Embedded IP Phone Display
- IP Phone Keep Alive
- IP Phone Failover
- Public Switched Telephone Network (PSTN) Failover
- Distributed CDR

2.6.1 Distributed Switch Control

The ShoreWare Telephony Management Service (TMS) runs on every ShoreWare distributed server, ensuring switch control even if there a WAN outage between the remote server and the headquarters site. Since multiple servers share the task of switch management, if a server fails, only the extensions it controls may be affected by a disruption in service.

2.6.2 Embedded IP Phone Display

The Embedded IP Phone Display feature essentially shifts support of several tasks related to IP phone operation from the server to the switch. This enhances system reliability and offers better uptime. The following features are supported on the switch and thus will continue to be available even when the server is down:

- Phone display
- Transferring a call
- Conference calls
- Placing calls on hold
- On-hook dialing
- Intercom
- Redial
- Pickup
- Park
- Unpark

The following features that require writing to the database will continue to be supported by the server and not the switch:

Directory
Options
Speed dial (due to its reliance on the database)
Ability to change call handling modes
Workgroup Agent Wrap up
Monitoring extensions on other switches
Presence information for user serviced by other switches

2.6.3 IP Phone Keep Alive

ShoreGear 1U Half-Width and 1U Full Width voice switches send a heartbeat to their associated IP phones once a minute. If the heartbeat is not acknowledged within approximately four seconds, the switch considers the IP phone to be offline or unavailable. The switch continues to broadcast the heartbeat every minute. Any currently offline IP phone that returns an acknowledgement is considered online and available.

2.6.4 IP Phone Failover

IP phones can be optionally configured to send a heartbeat to their ShoreGear switch every four minutes. If an IP phone cannot communicate with its switch, the phone automatically connects to another switch located at the same site. For IP phone failover to be effective, the system must be planned with sufficient excess capacity to handle phones from at least one switch during a failover event. For example, if a switch with 20 IP phone ports fails, 20 IP phone ports need to be available elsewhere at the same site.

The feature allows an administrator to configure the system so there can be failover of phones from one switch to another in the case of a switch failure. The feature is not intended to provide failover for network outages (i.e. or lost connectivity between the headquarters server and a remote site). Some network outage scenarios may be handled by the ShoreTel failover solution.

Failover will occur on a phone by phone basis and will be driven by receiving RSIP from the phone. IPCS will not move phones other than the one sending RSIP. Each phone must initiate its own failover.

For configuration details, see the ***ShoreTel Administration Guide***.

2.6.5 Public Switched Telephone Network (PSTN) Failover

User extensions can be optionally configured to route extension-to-extension calls to the public switched telephone network (PSTN) in the event that an IP connection is unavailable. Extension-to-extension calls are those a user makes to another site within a multi-site system, for example, a user in New York calling a co-worker at the company's San Francisco office. The IP connection may be unavailable due to lack of bandwidth or connectivity.

The PSTN failover option must be explicitly enabled for each user and bypasses the caller's call permissions. For configuration details, see the ***ShoreTel Administration Guide***.

For systems using the Distributed Routing Service (DRS), PSTN failover for outbound calls will not function when the local switches lose connectivity to a DRS server. When a site does not have connectivity to DRS, users at other sites with DRS connectivity will be able to reach the users at that site using PSTN failover (as long as the destination site has trunks to accept the PSTN calls). This limitation has the biggest impact for small offices that do not have a local ShoreWare server.

2.6.6 Distributed CDR

In the event of a WAN outage, Call Detail Record (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 Server's database. After two hours, the distributed server deletes the data and logs an error to the local server's NT event log.

2.7 Integrated Applications

The ShoreTel system includes a suite of applications that are integrated with the system. These applications (which are discussed in the following sections) include:

Account Codes
Voice Mail
Automated Attendant
Hunt Groups
Workgroups
Pickup Groups
Queue Monitor
Agent Monitor
Directory Viewer
History Viewer
Call Detail Recording
Desktop Call Control Service
Unified Messaging Service

TAPI-compliant, third-party applications can also be added on a distributed server. Such servers should have no voice mail users.

2.7.1 Account Codes

An Account Codes Collection Service (ACC) allows assignment of account codes or activity codes to outbound calls. The system supports account codes that can vary in length and format. Account code collection is enabled on a per-user group basis with the collection of account codes set to one of three states: disabled, optional, or forced. Call Detail Reports include details of the account codes associated with outbound calling. The Account Codes Service is associated with a configurable extension and has a dedicated user group that defines ultimate call permissions and trunk group access. In addition, wildcard characters can be used in place of any DTMF digit 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, with the introduction of wildcards, a length check is performed. The wildcard allows the system to support far more than the previous limit of 50,000 codes.

2.7.2 Voice Mail

The integrated voice mail application provides automated call answering, voice mail recording, and message playback. Since voice mail is simply a software application, there are no “port” or “storage” limitations as in traditional voice mail systems. To reduce WAN bandwidth utilization, the voice mail application can be distributed across the IP network.

Each mailbox supports five call handling modes (including Standard, In a Meeting, Out of the Office, Extended Absence, and Custom), each with its own greeting. Each mailbox also provides message notification to an extension, external number (cell phone), or pager.

Find Me forwarding and Auto Find Me forwarding allow calls to be forwarded from the voice mail greeting to up to two numbers. If the call is not accepted at either of the Find Me destinations, the call is returned to voice mail.

The Auto-delete by Number of Days feature allows a system administrator to set a maximum time limit, (ranging from a month to several years) for the storage of voice mail messages. The tool can be used to encourage users to better manage their voice mailboxes. When the feature is enabled and a user has old messages that are approaching the expiration time limit, the user will receive warning messages indicating that those voice mail messages will be deleted.

The Mailbox Full Notifications feature lets users know when their mailbox has approached the maximum capacity. 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 the notification 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.

The Voice Mail server has a limit of 200 deleted messages per mailbox. Deleted messages are not counted against the total message count. However, maximum number of deleted messages allowed at any point in time remains at 200 to limit total server mailbox storage space. If this limit is reached, deleting further messages will result in purging of the oldest deleted messages, so as to keep the count of the deleted messages to 200.

Escalation Notification is a traditional voice mail feature that allows support groups to offer round-the-clock service to their customers. Thus, if a customer calls into the ShoreTel voice mail system to leave a message requesting urgent service, the system will send out a page, phone call, or email to an employee in the support department. If this first employee were to ignore the notification for a specified period of time, another employee in the escalation profile would be contacted until someone listens to the customer's voice mail and handles the problem.

For details on configuring any of these voice mail features, please refer to the ***ShoreTel Administration Guide***.

For specific information about the supported capacity for voice mail on the ShoreTel system, see Table 2-2 on page 42.

2.7.3 Automated Attendant

The integrated automated attendant application provides automated call answering and call redirection, including dialing by name and dialing by number. As with voice mail, there are no “port” limitations such as exist in traditional systems. The automated attendant application is distributed across all the application servers when multiple servers are provisioned. All menus are available locally at every server. Calls directed to the automated attendant at a site with a server are handled by the local server.

Each automated attendant (AA) menu supports up to four different modes (On-Hours, Off-Hours, Holiday, and Custom) that can be automatically driven by schedules. In addition, users can record AA menu prompts from their own telephone, instead of having to go through ShoreWare Director. This ability frees the system administrator from having to be involved with the task of recording AA menus, allowing him or her to delegate the task to more appropriate team members.

Users can record a different AA menu prompt for each call-handling mode (On-Hours, Off-Hours, Holiday, and Custom). This feature can be enabled or disabled on a per-menu basis.

Each AA menu will have its own password and a unique, dialable number. A separate "Menu Mailbox" is created for each AA menu, allowing users to dial into the system to change the menu prompts in the same way that they would change their personal mailbox greeting.

For specific information about the supported capacity for automated attendants on the ShoreTel system, see Table 2-2 on page 42.

2.7.4 Hunt Groups

Hunt groups allow you to route calls to a list of extensions. Hunt groups can be accessed via an extension, DID, and/or DNIS mappings. Hunt groups are supported by ShoreGear switches and remain available when connectivity to the Headquarters server is lost. The hunt group can be used as the backup destination for a workgroup, so that some basic hunting can be done even when the workgroup server is not reachable. To maximize reliability, assign hunt groups to a switch close to the majority of the members and/or trunks associated with the hunt group.

A maximum of 8 hunt groups can be assigned to a single switch. A total of 16 user numbers can be assigned to hunt groups on a single switch.

For more information on hunt groups, see Section 12.10 on page 171.

2.7.5 Workgroups

The ShoreTel system provides the contact center with flexibility for distributing callers to available agents, as well as options for managing calls when agents are not available. Inbound calls are directed to a workgroup application on the headquarters server that distributes calls to agents in one of four administrator-configured patterns (Top Down, Round Robin, Longest Idle, or Simultaneous Ring). When no agents are available, calls can be directed to a queue where they are held until an agent becomes available.

Workgroup overflow and interflow capabilities can be configured to reduce the wait time for callers who are dialing into an ACD, thus ensuring faster service and greater customer satisfaction. ("Overflow" refers to transferring a call from one workgroup queue to another once a wait-time threshold has been exceeded, and "interflow" refers to transferring a call to another dialable number (e.g. an extension, menu, or an external number) once a wait-time threshold has been exceeded.

Alternatively, if calls are unanswered, they can be directed to a workgroup mailbox accessible by all agents. Agents may belong to multiple workgroups, and an agent's login status applies to all the workgroups of which that agent is a member.

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 this mode, agents remain logged in but do not receive new calls until the configured wrap-up time passes. This enables agents to complete any required updates to the customer records between calls.

When an agent is a member of multiple workgroups, and calls are available from different workgroups, the agent receives the longest waiting caller regardless of workgroup.

Each workgroup and each queue supports four different modes (On-Hours, Off-Hours, Holiday, and Custom) that can be automatically driven by schedules.

For specific information about the supported capacity for workgroups on the ShoreTel system, see Table 2-2 on page 42. For more information about configuring overflow and interflow, please see the "Configuring Workgroups" chapter in the ***ShoreTel Administration Guide***.

2.7.6 Pickup Groups

Pickup Groups are a traditional PBX and key system feature used in group environments to allow users in a pickup group to answer any ringing phone in that group. The feature works best in places where a several people work together on a daily basis, such as design firms. If a group member is away from her desk and across the room while her phone rings, she can quickly answer the call from another IP phone by pressing the relevant soft key or programmable button, or by using a simple (feature code) star command from an analog phone.

Pickup groups can include the following types of extensions:

- User extensions
- Workgroup extensions
- Bridged Call Appearance (BCA) extensions
- Extension Assignment extensions

For more information on configuring Pickup Groups, please refer to the ***ShoreTel Administration Guide***.

2.7.7 Queue Monitor

The ShoreWare Queue Monitor is embedded in the Agent and Supervisor ShoreTel Communicator client software. The Queue Monitor allows agents and supervisors to monitor business-critical queue statistics and information in real time.

For agents belonging to multiple workgroups, the Queue Monitor displays queue information for all workgroups of which the agent is a member.

2.7.8 Agent Monitor

The ShoreWare Agent Monitor provides workgroup supervisors with a real-time view on call center activity. The Agent Monitor shows status information for agents in all the workgroups of which the supervisor is a member, including the agent's login state (logged in, logged out, in wrap-up mode), current call activity, and current call duration.

2.7.9 Directory Viewer

Directory Viewer is a convenient phone book of system and personal contacts for anyone who does not use Microsoft Outlook. Users can view contacts, change contact information, and initiate calls from the viewer.

2.7.10 History Viewer

Available through ShoreTel Communicator, the History Viewer displays a detailed log of both incoming and outgoing calls. Users can search the history for phone numbers of past callers. For each call, the History Viewer displays the source or destination number, the start time, and duration.

2.7.11 Call Detail Record (CDR)

The ShoreTel system tracks all call activity on the system, across all locations, and generates call detail records into a single database on the main ShoreWare server. The system comes bundled with the reports that use information from the database, including User Activity, Trunk Activity, Workgroup Agent Activity, Workgroup Queue Activity, Workgroup Service Level Summary, Account Code Activity, and WAN Activity.

Web-based CDR reports offer the primary method of accessing and viewing CDR data in the MySQL database. Reports can be run from ShoreWare Director, and after the reports have been generated, they can be printed, exported, and navigated interactively. In addition, by purchasing the proper keyed license, users can run a web-based CDR report remotely from clients other than the headquarters machine.

The system also stores call information in a text file that can be used by third-party call accounting packages. And for the benefit of legacy call accounting systems that cannot read from a database or from a text file, the ShoreTel system supports the ability to send CDR data out a serial port on the main ShoreWare server. If the serial port should become unavailable, the CDR data will be queued in a buffer for 300 seconds to help prevent the loss of data.

To make it easier for the ShoreTel system to integrate with various third-party SNMP monitoring tools, the ShoreTel system formats CDR media stream statistics and stores the data in a log file on the system. This helps users acquire a more accurate picture of the traffic patterns in their network, and the information can be useful in performing load analysis, identifying peak traffic times, and assisting the customer in setting up competitive pricing strategies.

2.7.12 Desktop Call Control Service

The Desktop Call Control service application provides call status and call control to every user on the system. This is provided through a Remote TAPI Service Provider (RTSP) that is on every desktop using ShoreTel's Communicator applications and other desktop TAPI applications.

2.7.13 Unified Messaging Service

Unified Messaging, ShoreTel's Outlook integration feature, provides an interface to the messaging applications on the desktop computers. This feature provides access to voice mail from Microsoft Outlook for each user, enabling users to manage their voice mail messages in the same way that they currently manage their e-mail messages.

In addition, Unified Messaging enables access to the system directory and each user's personal options. Unified Messaging also allows users to take advantage of the calendar-based call handling feature, which lets employees customize how calls are routed when they are not available.

2.8 Optional Applications

To augment the ShoreTel solution, ShoreTel offers conference bridge and contact center applications as system options.

2.8.1 ShoreTel Converged Conference Solution

The ShoreTel Converged Conference Solution provides easy-to-use, cost-effective audio-and-data conferencing. During a conference call, users can share PowerPoint, MSWord, Excel, or text documents with other conference participants.

The ShoreTel Converged Conference Solution includes the ShoreGear Converged Conference Bridge, ShoreTel Conference Manager, and ShoreTel Conference Director. The ShoreGear Converged Conference Bridge is a 1U rack-mounted conference server connected to your ShoreTel system via an Ethernet connection to your IP network. The ShoreTel Conference Manager is an intuitive, browser-based interface for conference call scheduling and call control. ShoreTel Conference Director is a browser-based management interface for the administration and maintenance of the ShoreGear Converged Conference Bridge.

2.8.1.1 ShoreGear Converged Conference Bridge

The ShoreGear Converged Conference Bridge is an embedded, preconfigured conference appliance that interfaces to your ShoreTel system via your IP network. The conference bridge supports 12, 24, 32, 48, or 96 ports.

2.8.1.2 ShoreTel Conference Manager

The ShoreTel Conference Manager enables conference call users to:

- Establish reservationless conferences
- Set up scheduled and recurring conference calls
- Start a conference call "on the fly"

For more information on ShoreTel Conference Manager, see the *ShoreTel Conference Manager User Guide*.

2.8.1.3 ShoreTel Conference Director

ShoreTel Conference Director provides an intuitive interface for operations, administration, maintenance, server configuration, service/user provisioning, and monitoring/alarm control.

Required authorization and authentication ensures that only valid users use the conference bridge services. To meet the highest security requirements, the server utilizes SSL encryption for secured messages and server side digital certificates.

2.8.2 ShoreTel Contact Center Solution

ShoreTel Contact Center Solution is a comprehensive routing and management system designed to control and monitor the activities of your contact center. The ShoreTel Contact Center Solution includes the ShoreTel Contact Center Server Software or ShoreTel Enterprise Contact Center Server Software, ShoreWare Contact Center Director, ShoreWare Agent Manager Software, and ShoreWare Agent Toolbar Software.

The ShoreTel Contact Center Server Software, together with its Interactive Voice Response package (IVR), provides the contact center administrator with sophisticated call routing options. These options include routing incoming calls by customer ID (or ANI), routing incoming calls by DNIS (the number dialed), routing incoming calls according to the agent that best fits the skill required (skills-based routing), statistical routing to route the incoming call by TSF (Target Service Factor), and more. In addition, the ShoreTel Contact Center Server Software uses scripts to collect information from the organization's database and the callers, using many IVR actions, and routes the call according to that information.

Incoming calls are routed to agents according to:

- The service required by the DNIS (number dialed)
- The customer, if the customer is identified in the organization's database
- A call control script that directs calls as specified by the caller
- Best skill fit of the agent
- The longest waiting time

The ShoreWare Contact Center Director module enables authorized supervisors to define the parameters of different system entities (for example, agents, agent groups, trunk groups) and easily modify their profiles. There are several administration levels with different access rights.

The ShoreWare Agent Manager monitors contact center activities and provides real-time information, as well as generates reports summarizing the system performance over a given time period. The ShoreWare Agent Manager also provides statistical analysis of the contact center system behavior within a specified period.

The ShoreWare Agent Toolbar provides the agent with all the necessary information regarding the type of an incoming call and caller, before the agent answers the call. Agents can perform all telephony functions from their desktops with this Windows-based application.

2.8.3 ShoreWare System Monitor

ShoreWare System Monitor is a Windows 2003/XP service that uses SNMP to monitor statistics and utilization for each interface on each switch. If data-link errors or utilization rates rise above a settable threshold, the generated web pages help determine the source of the network problems.

ShoreWare System Monitor discloses network weaknesses that cause data and VoIP stability issues, and by monitoring all network interfaces for utilization, packet loss, and errors, it becomes easy to determine exactly where network faults exist.

ShoreWare System Monitor provides information about the specific error or issue that is causing degradation to assist in troubleshooting and resolution, and it maintains a history of utilization and errors on all interfaces to assist in troubleshooting VoIP and network problems after they occur.

All network devices that support SNMP can be queried for link status and health information

2.9 Desktop Applications

ShoreTel provides a suite of integrated desktop productivity applications targeted at the needs of different users. The ShoreTel Communicator applications offer varying levels of functionality suited to different role requirements.

Refer to the ShoreTel Communicator User's Manual for more information.

2.9.1 Communicator for Web

Users can have remote access to their call handling options via a browser-based interface.

2.9.2 Extension Assignment

Extension Assignment allows users to maintain an on-system extension presence at an external PSTN number. A mapping is created between a user's office phone extension and his cell phone or PSTN phone number (at his home office), making it appear as though his PSTN phone is part of the ShoreTel system. The feature allows the user to manage the call via ShoreTel Communicator, so while the conversation occurs over the cell phone or home phone, the call appears via ShoreTel Communicator and can be acted upon using many of the features available via ShoreTel Communicator.

2.10 Voice Switches

The ShoreGear voice switches provide the physical connectivity to voice endpoints and provide a highly reliable, highly scalable platform for the ShoreWare distributed call control software. The call control software runs on top of VxWorks, a real-time embedded operating system designed specifically for mission-critical applications. The voice switches have FLASH memory that allows permanent storage of the call control software and configuration information. Except for a highly reliable fan, the voice switches have no moving parts (for example, no hard drive). Internal sensors automatically monitor the fan as well as the temperature, and if any failure occurs the system can automatically notify the system administrator, through e-mail if desired. The voice switches include the necessary Digital Signal Processor (DSP) technology to enable toll-quality voice, with features such as echo cancellation, voice compression, and silence suppression.

Each ShoreGear voice switch connects to the IP network using a 10/100M Ethernet interface. If more sites or ports are required, you simply connect additional ShoreGear voice switches to your IP network. The system is inherently scalable, unlike legacy PBX systems that have hardware growth limitations with line cards, shelves, cabinets, and systems.

ShoreGear voice switches reboot in less than 60 seconds, providing fast fault recovery. ShoreGear voice switches feature a backup operator in case the site operator is unreachable due to network outages.

Refer to Appendix G, starting on page 303 for a complete description of all ShoreGear Switches.

2.11 ShoreTel IP Phones and Devices

Both analog and IP telephones are available from ShoreTel. With ShoreTel IP phones, you create an end-to-end IP network, or a single-wire-to-the-desktop solution. The ShoreTel IP phone’s intuitive user interface gives the user a high comfort level when performing phone operations.

For specific information about the supported capacity for IP and analog telephones on the ShoreTel system, see Table 2-2 on page 42.

2.11.1 ShorePhone-AP100

The ShorePhone-AP100 telephone provides a cost-effective analog solution for business desktops.

Key features include:

- Large display for caller name, number, and directory access
- High-quality speaker telephone
- Menu access to common features

2.11.2 ShorePhone-IP110

The ShorePhone-IP110 phone is a cost-effective phone designed for general use.

Key features include:

- Single-line display for caller information
- Six function keys (Conference, Hold, Intercom, Redial, Transfer, Voice Mail)
- Ethernet Switch port for connecting a PC to the back of the phone
- Support for basic media encryption for calls inside a ShoreTel network
- Ability to load custom ring tones in .wav file format

2.11.3 ShorePhone-IP115

The ShorePhone-IP115 phone is a cost-effective phone that is based on the IP110 model, but with the addition of an external microphone to support speakerphone functionality.

Key features include:

- External microphone to support speakerphone
- Single-line display for caller information
- Six function keys (Conference, Hold, Intercom, Redial, Transfer, Voice Mail)
- Ethernet Switch port for connecting a PC to the back of the phone
- Support for basic media encryption for calls inside a ShoreTel network
- Ability to load custom ring tones in .wav file format

2.11.4 ShorePhone-IP212k

The 212k IP phone is designed to function as a key phone and offers 12 custom buttons that can be used for line appearance and other functions. The 212k is ideal for small offices and branch offices that require key system functionality.

Key features include:

- Scrolling text that allows for the display of more information.
- Menu and Select buttons that provide services similar to soft keys and scroll bar, and that assist in phone navigation and programming
- Eight function keys (Voice Mail, Transfer, Options, Conference, Directory, Intercom, Redial, Hold)
- InstaDial™ functionality in which calls are automatically transferred after digit collection stops and a configurable timeout period has expired
- Automatic Off-Hook Preference that lets users select which audio path. (speakerphone or headset) is auto-activated when calls are sent or received
- Ethernet Switch port for connecting a PC to the back of the phone
- A first (upper left-most) custom button that is reserved for line appearance only and cannot be configured to perform other functions
- Support for basic media encryption for calls inside a ShoreTel network.
- Ability to load custom ring tones in .wav file format
- Built-in handset lifting functionality to support certain Plantronics wireless headset models

2.11.5 ShorePhone-IP230

The ShoreTel IP230 Phone is a full-featured yet inexpensive IP phone that is similar to the ShoreTel IP210 phone, but with the added functionality of programmable buttons.

Key features include:

- The IP230 has 3 custom buttons that can be programmed for extension monitoring, speed dial, and other functions. Note that the top-most button is reserved for line appearance. (Each “button” is an LED-enabled hard key and has an associated 6 character label on the LCD.)
- Eight function keys (Voice Mail, Transfer, Options, Conference, Directory, Intercom, Redial, Hold)
- InstaDial™ functionality in which calls are automatically transferred after digit collection stops and a configurable timeout period has expired
- Monochrome black and white display
- Automatic Off-Hook Preference that lets users select which audio path (speakerphone or headset) is auto-activated when calls are sent or received
- Ethernet Switch port for connecting a PC to the back of the phone
- Support for basic media encryption for calls inside a ShoreTel network
- Support for two-line caller ID display feature, which displays the caller name and number on two separate lines for in-coming calls and outbound calls
- Ability to load custom ring tones in .wav file format
- Built-in handset lifting functionality to support certain Plantronics wireless headset models

2.11.6 ShorePhone-IP265

The new ShoreTel IP265 Phone is a full-featured yet inexpensive IP phone that is similar to the ShoreTel IP230 phone but with an additional three programmable buttons (for a total of six, as opposed to three on the IP230 model) and a color LCD display.

Key features include:

- 2.7-inch TFT-LCD Color display with backlighting
- Ability to download and display a 24-bit .bmp “wallpaper” file
- Six custom buttons that can be programmed for extension monitoring, speed dial, and other functions. Note that the top-most button is reserved for line appearance. (Each “button” is a tri-color LED-enabled hard key and has an associated 6-character label on the LCD.)
- Eight function keys (Voice Mail, Transfer, Options, Conference, Directory, Intercom, Redial, Hold)
- InstaDial™ functionality, in which calls are automatically transferred after digit collection stops and a configurable timeout period has expired
- Automatic Off-Hook Preference that lets users select which audio path (speaker, headset, wireless headset) is auto-activated when calls are sent or received
- Ethernet Switch port for connecting a PC to the back of the phone
- Support for basic media encryption for calls inside a ShoreTel network
- Support for two-line caller ID display feature, which displays the caller name and number on two separate lines for in-coming calls and outbound calls
- Ability to load custom ring tones in .wav file format
- Built-in handset lifting functionality to support certain Plantronics wireless headset models

2.11.7 ShorePhone-IP560

The ShorePhone-IP560 is a high-end phone designed for executives, assistants, and operators who handle high call volumes and share call flows with other users. Key features include:

- Caller ID display for up to six calls simultaneously
- Backlit display
- Monitoring support for up to five extensions
- Eight function keys
- Four soft keys
- Ethernet Switch port for connecting a PC to the back of the phone
- InstaDial™ functionality in which calls are automatically transferred after digit collection stops and a configurable timeout period has expired
- Automatic Off-Hook Preference that lets users select which audio path (speakerphone or headset) is auto-activated when calls are sent or received
- Support for basic media encryption for calls inside a ShoreTel network
- Support for programmable buttons and extension monitoring
- Support for two-line caller ID display feature, which displays the caller name and number on two separate lines for in-coming calls and outbound calls
- Ability to load custom ring tones in .wav file format
- Built-in handset lifting functionality to support certain Plantronics wireless headset models

2.11.8 ShorePhone-IP560g

The ShorePhone-IP560g is a high-end phone designed for executives, assistants, and operators who handle high call volumes and share call flows with other users, yet require the benefit of 1000BaseT operations.

Key features include:

- Supports 10BaseT, 100BaseT, and 1000BaseT operations
- Six custom buttons that can be used for line appearance and other functions
- Eight function keys (Voice Mail, Transfer, Options, Conference, Directory, Intercom, Redial, Hold)
- Four soft keys
- Gigabit Ethernet Switch port for connecting a PC to the back of the phone
- Caller ID displayed for up to six calls simultaneously
- Backlit display
- Monitoring for up to five extensions
- InstaDial™ functionality in which calls are automatically transferred after digit collection stops and a configurable timeout period has expired
- Automatic Off-Hook Preference that lets users select which audio path (speakerphone or headset) is auto-activated when calls are sent or received
- Support for basic media encryption for calls inside a ShoreTel network
- Support for programmable buttons and extension monitoring
- Support for two-line caller ID display feature, which displays the caller name and number on two separate lines for in-coming calls and outbound calls
- Ability to load custom ring tones in .wav file format
- Built-in handset lifting functionality to support certain Plantronics wireless headset models

Installation Notes:

The ShoreTel IP560g telephone requires a gigabit-compatible Power over Ethernet (POE) power supply that complies with IEEE802.af. The 560g phone is a Class 3 device with a maximum consumption of 8.2 watts. Please use 8.2 watts for capacity planning with Gig POE switches on multiple deployments.

The 560g model requires more power than the other ShoreTel models, and thus the 560g phone is not compatible with the ShorePhone power adapter used with other ShoreTel phone models.

The 560g model cannot be daisy-chained from the Button Box (BB24). The BB24 passthrough power is limited to Class 2 devices and the 560g is a Class 3 device. This means the BB24 cannot forward adequate power to an IP 560g phone.

The ShoreTel IP560g telephone requires the use of Category 5e or Category 6 Ethernet cables. Using Category 5 Ethernet cables is not officially supported and may lead to lower connection speed and/or performance issues during high-data transfer scenarios.

2.11.9 ShorePhone-IP565g

The ShorePhone-IP565g is a high-end phone designed for executives, assistants, and operators who handle high call volumes and share call flows with other users, yet require the benefit of 1000BaseT operations, and who want the ability to use a Bluetooth wireless headset.

Key features include:

- 3.5-inch TFT-LCD Color display with backlighting
- Ability to download and display a 24-bit .bmp “wallpaper” file
- Support for some Bluetooth wireless headset models
- Supports 10BaseT, 100BaseT, and 1000BaseT operations
- Six tri-color custom buttons that can be used for line appearance and other functions
- Eight function keys (Voice Mail, Transfer, Options, Conference, Directory, Intercom, Redial, Hold)
- Four soft keys
- Gigabit Ethernet Switch port for connecting a PC to the back of the phone.
- Caller ID displayed for up to six calls simultaneously
- Monitoring for up to five extensions
- InstaDial™ functionality in which calls are automatically transferred after digit collection stops and a configurable timeout period has expired
- Automatic Off-Hook Preference that lets users select which audio path (speaker, headset, wireless headset, or Bluetooth) is auto-activated when calls are sent or received
- Support for basic media encryption for calls inside a ShoreTel network
- Support for programmable buttons and extension monitoring
- Support for two-line caller ID display feature, which displays the caller name and number on two separate lines for in-coming calls and outbound calls
- Ability to load custom ring tones in .wav file format
- Built-in handset lifting functionality to support certain Plantronics wireless headset models

Installation Notes:

The ShoreTel IP565g telephone requires a gigabit-compatible Power over Ethernet (POE) power supply that complies with IEEE802.af. The 565g phone is a Class 3 device with a maximum consumption of 8.2 watts. Please use 8.2 watts for capacity planning with Gig POE switches on multiple deployments.

The 565g model requires more power than the other ShoreTel models, and is thus not compatible with the ShorePhone power adapter used with other ShoreTel phone models.

The 565g model cannot be daisy-chained from the Button Box (BB24). The BB24 passthrough power is limited to Class 2 devices and the 565g is a Class 3 device. This means the BB24 cannot forward adequate power to an IP 565g phone.

The ShoreTel IP565g telephone requires the use of Category 5e or Category 6 Ethernet cables. Using Category 5 Ethernet cables is not officially supported and may lead to lower connection speed and/or performance issues during high-data transfer scenarios.

2.11.10 ShorePhone-IP655

ShoreTel IP Phone 655 is ShoreTel's flagship IP phone designed for executives, executive assistants and for conference room use. The ShoreTel IP Phone 655 is a new phone that offers the highest quality performance and enhanced IP telephony functionality.

ShoreTel IP Phone 655 provides a touch screen user interface that is very intuitive and enhances user productivity. Advanced applications enaled by the large color touch screen display include enhanced Directory and Call history applications with real-time telephony presence information, an a new visual Voicemail with both playback and composition capabilities. For more details about the ShoreTel IP Phone 655, see ShoreTel IP Phone 655 Installation and Users Guide.

Key features include:

5.7" inch 640 x 480 pixel color touch screen display with haptic feedback6 microphone beam-forming array for advanced speakerphone acoustics

Connectors for up to two remote microphones for better conference room coverage

10/100/1000 Ethernet

Five capacitive touch keys (mute, speakerphone, headset, redial, and volume control)

2-port Ethernet Switch for connecting a PC to the network through the phone.

Customizable wallpapers and ring tones configured with Director

Twelve programmable virtual buttons that can be used for line appearance, speed dial, extension monitoring, and other functions

Built-in electronic headset lifter functionality compatible with some models of Plantronics wireless headsets.

Visual Voicemail with message review and composition capabilities

Call History with live telephony presence and call categorization

Directory with live telephony presence, first/last name sortability, and touch screen virtual keyboard for improved filtering

Ability to handle up to sixteen simultaneous calls

Ability to monitor up to eleven other extensions

InstaDial™ functionality in which calls are automatically transferred after digit collection stops and a configurable timeout period has expired

Automatic Off-Hook with user selectable audio path (speaker, headset, or wireless headset) when calls are dialed or answered

Support for media encryption for calls to other ShoreTel IP Phones and trunks

Support for caller ID display of name and number as well as Trunk Group and Workgroup information for certain calls

Large, readable fonts

Installation Notes:

The ShoreTel IP655 telephone requires Power over Ethernet (POE) power supply that complies with IEEE802.af. The 655 phone is a Class 3 device with a maximum consumption of 8.2 watts. Please use 8.2 watts for capacity planning with Gig POE switches on multiple deployments.

The 655 model requires more power than the other ShoreTel models, and is thus not compatible with the ShorePhone power adapter used with other ShoreTel phone models.

The ShoreTel IP655 telephone requires the use of Category 5e or Category 6 Ethernet cables. Using Category 5 Ethernet cables is not officially supported and may lead to lower connection speed and/or performance issues during high-data transfer scenarios.

2.11.11 ShorePhone-BB24

The ShoreTel 24 IP Button Box provides additional shortcut functions for users of the multiline phones. The BB24 behaves like an additional set of 24 custom buttons in additional to the buttons that already exist on the multiline phones.

Key features include:

Twenty-four custom keys
Ability to assign up to 4 Button Boxes to a multiline phone
Support for Programmable Buttons feature
Ability for each user to define layouts for up to four BB24's thus allowing a maximum of about 100 programmable buttons for most phones (exact number varies depending on which phone the BB24 is connected)
Custom buttons in which each is an LED-enabled hard key and has an associated 6 character label on the LCD
Ethernet Switch port for connecting a PC to the back of the phone
Ability to forward power to one additional unpowered device to support a daisy-chain configuration

For detailed information on available options and how to use them, refer to the *ShoreTel Programmable Buttons User Guide*.

For installation instructions, refer to the *ShoreTel 24 IP Button Box Quick Install Guide*.

2.12 System Capacity

The ShoreTel Release 11 system can scale incrementally up to 10,000 ports (users and/or trunks) representing 500 ShoreGear voice switches over the entire system. The system is completely nonblocking and can support 5,000 simultaneous calls at a rate of 50,000 calls per hour (depending upon server configurations).

Table 2-2 provides a summary of the ShoreTel system capacity

Component	Capacity	Notes
System		
Sites	500	Exact number varies by configuration.
Switches	100/site 500/system 100/server	Exact number varies by configuration.
Route Points	300/server	This is per server
Analog Ports	5,000	Exact number varies by configuration.
IP Phones	10,000 (max)	Exact number varies by configuration. See Server capacity table.
Simultaneous Calls	5,000	5,000 calling 5,000.
Busy Hour Call Completion	50,000	Depending upon server configurations
Users		
Users	10,000	
– Port Based Users	5,000	
– IP Phone Users	10,000	
– Virtual Users	1,000/server	
User Groups	250	
Telephony Permissions	100	
Call Permissions	100	
Voice Mail Permissions	100	
Trunks		
Trunks	5,000	
Trunk Groups	250	
Number of Trunks/TG	500	
Servers		
Number of servers	21	1 main, 20 distributed (for voice mail, auto-attendant, messaging, directory, configuration services, and desktop call control). Each server is certified to support up to 1,000 users.
Number of VMBs	100	
Number of 3rd Party SIP Servers	20	
Media streams (G.711 per server)	254	Simultaneous voice mail sessions, for example.
Media streams (G.729 per server)	Media streams (G.729 per server)	
Media streams (total)	6,234	21 servers x 254 media streams per server. Workgroups can exisit now in any DVS + 100 VMB x 9 streams per VMB.
Voice Mail		
Mailboxes (total)	10,000	These can be distributed across the servers.
Mailboxes (per server)	3,000	

Table 2-2 ShoreTel System Capacity

Component	Capacity	Notes
Storage	Unlimited	Restricted by the size of disk available (1 hour of voice mail per 30 MB of disk storage).
Auto-Attendant		
Menus (total)	1,000	Every server has every menu.
Hunt Groups		
Hunt groups per switch	8	
Total hunt group members per switch	16	
Workgroups		
Workgroups (total)	128	
Members per workgroup	300	Top down, round robin, and longest idle hunt pattern.
WG Agents (total per system)	300	
WG Agents	16	Simultaneous ring.
Calls in Queue per Queue	254/server	Overflow is directed to the workgroup backup extension.
BHCC/system without reports during business hrs	Large HW = 10K / Med HW = 5K / Small HW = 1K	See Server HW specs for size & traffic considerations
BHCC/system with reports during business hrs	Large HW = 5K / Med HW = 1K / Small HW = not supported	See Server HW specs for size & traffic considerations
Max # of PCM's in WG server	300	
Paging Groups		
Paging Groups (total)	300/system	
Paging Group Members	300/system	
Max # of simultaneous pages	100/server	
Account Code		
Account Code (per system)	Account Code (per system)	
Call Detail Record		
Storage	1.5 GB (MySQL has a capacity of 64TB)	500,000 workgroup calls, OR 1.5 million extension-to-extension calls, OR 1.0 million combined call records Implementing a database of this size typically requires 4.0 GB of disk space, including 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).
ShoreTel Communicator		

Table 2-2 ShoreTel System Capacity

Component	Capacity	Notes
ShoreTel Communicators (total)	10,000	
ShoreTel Communicators (per server)	1,000	
Personal	10,000	
Professional	10,000	
Workgroup Agent	300	
Workgroup Supervisor	128	
Workgroup Agent/server	300 per server, 300 per system	
Workgroup Supervisor/ server	128 per server, 128 per system	
Operator	200	250 monitored extensions maximum.
ShoreTel Communicator for Mobile	1,000	Per system.
Music on Hold (MOH)		
Music on Hold (MOH)	15	One switch can provide MOH for up to 15 switches per site.
Programmable Buttons		
IP phone buttons configured for extension monitoring (per switch)	1024	
Phones that can monitor an extension	32	
Voice Switch Capacity		
Media streams/switch (No encryption)	60	
Media streams/switch (encryption)	60	
Media streams/switch (SRTP)	40	
Media streams/switch (SRTP + authentication)	30	
G711 Limits for VMB	9	
G729 Limits for VMB	5	
BAA Simultaneous # of calls - Voice Switches	60	
Simultaneous # of calls SIP Ringing - Voice Switches - G711	60	
Simultaneous # of calls SIP Ringing - Voice Switches - G729	0	

Table 2-2 ShoreTel System Capacity

The following tables contain information on how to select a server for your ShoreTel implementation.

Server requirements are specified in 3 tiers: servers for small systems that support up to 500 users, servers for medium sized systems that support up to 2,500 users, and servers for large systems that support up to 10,000 users. The following table describes the key capacity limits for each of the new server tiers.

Size	Maximum number of users per System	Maximum number of users assigned per Server	Maximum System BHCC[1]	Maximum BHCC per server[2] Reports run outside business hours	Maximum BHCC per server[2] Reports run during business hours
Small	500	500	5,000	1,000	Not Recommended[3]
Medium	2,500	1,000	25,000	5,000	1,000
Large	10,000	1,000	50,000	10,000	5,000

Table 2-3 System and Service Capacities

NOTE: Busy Hour Call Completion (BHCC) includes all traffic that can occur in that server - regular voice calls, workgroup calls, voicemail etc.

[1]BHCC (Busy Hour Call Completion) per system is the total number of calls in the system during the busy hour including internal and external calls and including calls terminated to desk phones, softphones, trunks, or server applications such as voicemail.

[2] BHCC per server is based on the number of calls actually handled by the server during the business hour including workgroup calls in menus and queues, auto-attendant calls and calls to the voicemail service.

[3] The ShoreTel report generation tools that run on the server are configured by default to run at a lower priority than other, more critical services. A light demand of report generation should have little or no affect on a server with adequate minimum performance specifications. If you are a heavy report user or experience any degradation of voicemail or other server prompts on an underpowered server, you must move up to the next tier level of servers.

To select a server for your new system deployment, first consult the sizing table and determine the tier of the server needed using the system and per server specifications. Then match that size (small, medium, or large) to the server requirements below.

Size	Processor	RAM	Network
Small	Intel Core 2 Duo E8400, Single Dual Core 3.00 GHz or Intel® Core™ i3-540 Processor (4M Cache, 3.06 GHz)	4 GB	100 T-Base
Medium	Intel Xeon 5520 Single Quad Core 2.27 GHz	8 GB	100 Base-T or Gigabit Ethernet
Large	Intel Xeon 5520 Dual Quad Core 2.27 GHz	8 GB	Gigabit Ethernet

Table 2-4 Server Specifications

NOTE: The new hardware specifications are to be used to size servers running ShoreTel's Headquarters server software as well as ShoreTel's Distributed Voice Services software. For exmaple, consider a 2 location system with 2,000 users and 20,000 BHCC. A Headquarters server is located at the main site and a Distributed Voice Services server is located at the remote site. Each of the servers handle 2,000 BHCC. In this case, both servers should be provisioned with hardware that meets the medium tier of hardware requirements because the system capacity and both server capacities fall within this tier.

When deploying servers for medium or large systems, please note that you must select an operating system with support for expanded memory by using the Enterprise Editions of either Microsoft Windows 2003 or 2008 Server.

2.12.0.1 Busy Hour Call Completion (BHCC) and Busy Hour Call Attempts (BHCA)

Busy Hour Call Completion (BHCC) per system is the total number of calls in the system during the busy hour, including internal and external calls and including calls terminated to desk phones, softphones, trunks, or server applications such as voicemail. This includes all traffic that can occur in the server - regular voice calls, workgroup calls, voicemail etc.

Busy Hour Call Attempts (BHCA) is the number of calls attempted at the busiest hour of the day (peak hour). The higher the BHCA, the higher the stress on the network processors. If a bottleneck in the network exists with a capacity lower than the estimated BHCA, then congestion will occur resulting in many failed calls and customer dissatisfaction.

BHCC Limits for Server Tiers

Server requirements are specified in 3 tiers: servers for small systems that support up to 500 users, servers for medium sized systems that support up to 2,500 users, and servers for large systems that support up to 10,000 users.

Size	Maximum number of users per System	Maximum number of users assigned per Server	Maximum System BHCC	Maximum BHCC per server Reports run outside business hours	Maximum BHCC per server Reports run during business hours
Small	500	500	5,000	1,000	Not Recommended[3]
Medium	2,500	1,000	25,000	5,000	1,000
Large	10,000	1,000	50,000	10,000	5,000

BHCC per server is based on the number of calls actually handled by the server during the business hour, including workgroup calls in menus and queues, auto-attendant calls and calls to the voicemail service.

Call Load Capacity for Switches

A ShoreTel system supports a maximum of 100 Voicemail Model Switches. There are no restrictions concerning the allocation of switches among the sites defined by the system.

For ShoreGear Voicemail Model Switches, call load capacity is:

• 5400 BHCC when supporting 90 MGCP IP Phones or 90 SIP Trunks

• 3600 BHCC when supporting 90 SIP IP Phones or 90 SIP Trunks

BHCA Call Volume

The system supports 5,000 BHCA.

2.12.1 Extension Monitoring Limitations

Note that there is a limit to the number of extensions that can be monitored, whether from a ShorePhone-BB24 device or from a ShorePhone multiline phone. This limitation is dependent on two factors:

Update rate (every call causes one or more monitoring phones to be updated)
Whether the monitoring phones are spread across one or more switches

ShoreTel switches support an update rate of 1 per second. This limit is independent of whether the monitored extensions are on the same switch or a different switch. If the monitored extensions are on a different switch, then IPDS is involved.

2.12.2 ShoreGear Voice Switch Feature Capacity

The ShoreGear voice switch is designed to handle the maximum load for the services it provides. Some features place a higher real-time load on the ShoreGear voice switch processor than others, and the use of these features must be carefully planned to take into account the impact on the processing power of a switch to handle call control signaling messages.

Table 2-5 offers some general guidelines for the number of extensions and group members for several commonly used features. Keep in mind that in addition to observing these limitations, you must stay below the real-time requirements of the switch itself.

	Hunt Group	Bridged Call Appearance	Pickup Group	IP Phones
Extension	8	24	16	120
Members / extensions	16	32 – phones on a switch monitoring the same extension	24	N/A
Stack size/extensions	24	24	N/A	24
Total members on all extensions	16	N/A	80	N/A

Table 2-5 Feature Capacity

2.12.2.1 IP phones

Ringing a single user's IP phone is generates only one set of call control messages. However, as the call rate increases, the load on the processor also increases. Note that the call rate is the driving factor of load and not the length of a call. For instance, sixty calls placed over one hour, with each call lasting one minute, is a much higher load on the processor than a single call lasting one hour.

2.12.2.2 Hunt Groups

Hunt groups place a significantly heavier burden on the ShoreGear voice switch. For example, if you have a hunt group with 16 members, a single call into the hunt group will generate 16 simultaneous calls (assuming the feature is configured to simultaneously ring each hunt group member).

To extend this example, assume that the call stack size for this hunt group is set to 16, and 16 calls arrived at the same time, this would be equivalent to 256 calls (16 x 16) simultaneous calls. The number of hunt group members (as well as the call stack depth) is a multiplying factor for the signaling load that would be generated – thus, you should closely engineer hunt groups to ensure that the voice switch is not overburdened in order to ensure optimal performance.

You can have 8 hunt groups on a switch. Each hunt group can have up to 16 members and each hunt group can have a call stack of 24. The maximum number of members across all groups on the switch also has to remain below 16. For instance, you could have one hunt group of 16 members or 2 hunt groups with 8 members each.

2.12.2.3 Bridged Call Appearances

With Bridged Call Appearances (BCA), additional processor load is related to the call control signaling transmission to the buttons that have been programmed on the ShoreTel IP phones. If a single BCA with a call stack of one is configured on a phone, this represents one load. However, if that same BCA were to appear on 24 different phones, that would represent 24 times more call signaling load than if the BCA were to appear on one phone.

The switch is capable of handling 24 BCAs, with a call stack depth of 24 and up to 32 phones monitoring a single BCA. If there are no hunt groups on the switch, it is possible for the switch to support up to 160 buttons programmed to monitor BCAs.

2.12.2.4 Pickup Groups

Pickup Groups place an additional load on the processor related to tracking the extensions in the group (although the actual real-time load is rather light and is not factored into the real time equation).

The switch is capable of supporting 16 pickup groups with a maximum of 24 members in the group. The total number of members in all groups on the switch must not exceed 80.

2.12.2.5 Real Time Capacity

In addition to the overall feature capacity limit, you should calculate the real-time load on the switch using the formula below:

$$\left(\sum_{n=1}^{No\ HG} HG\_StackSize(n) * HG\_Members(n)\right) + (.5 * No.\_BCA\_Monitoring\_Lines) \le 80$$

Thus, with the following configurations:

a hunt group with four members and a call stack of four
a second hunt group with eight members and a call stack of three
ten phones, each monitoring four BCA

You would have room to spare:

HG 1	+	HG 2	+	BCAs	=	Total
4 x 4	+	8 x 3	+	(10 x 4)/2	=	60
16	+	24	+	20	=	60

CHAPTER 3

Planning and System Design

This chapter guides you through the initial design of your new voice communications system.

3.1 Checklist

The purpose of this chapter is to compile a high-level design of your system. The key components of the high-level design are:

Task	Description
Determine System Topology	page 49
Determine Telephone Requirements	page 51
Determine Trunk Requirements	page 53
Determine Number of ShoreGear Voice Switches	page 53
Determine WAN Connections	page 54

Table 3-1 Planning and System Design Checklist

3.2 Recommendations

The following recommendations will assist you in designing your new voice communications system.

- Make sure you understand all the unique routing and hunting requirements of your current system.
- Be sure to account for all devices, including conference rooms, lobby phones, fax machines, and modems.
- Make sure you consider the changes to the call flow and overall system design that may drive the need for additional trunks.

3.3 Determine System Topology

The ShoreTel system has a unique distributed call control software architecture that enables you to deploy ShoreGear voice switches and IP phones anywhere across your IP network. Even though multiple sites are supported, the ShoreTel system is a single system with an extensive set of integrated applications and a single management image. The ShoreTel system offers unmatched simplicity through this single image system, and delivers high availability, with no single point of failure, through its distributed architecture.

The first step in designing your voice network is to determine your overall network topology, which should provide the following information:

- **Sites and Users**. Number of sites and number of users at each site.

Headquarters and Distributed ShoreWare Servers. Number of servers required, plus the name or IP address of all ShoreWare servers (main and distributed).

Teleworker Sites. Number of teleworker installations and the type of telephones supported.

Telephone Requirements. Number of telephones at each site (by type).

Trunk Requirements. Number of trunks required for optimal performance.

ShoreGear Voice Switches. What models are needed and how many of each model.

WAN Connections. The number of WAN connections (per site) and complete service-level information.

See Chapter 9, starting on page 107, for detailed information on planning your network for the ShoreTel system.

3.3.1 Sites and Users

Your network topology diagram should provide an accurate inventory of the different physical sites and the number of users at each site.

3.3.2 Headquarters and Distributed ShoreWare Servers

The Headquarters ShoreWare server hosts the voice applications platform and the management web site, as well as the integrated voice applications. Typically, the Headquarters ShoreWare server is located at the largest location, containing the majority of users. Make special note of the main ShoreWare server on your topology diagram.

On your topology diagram, provide the following information about ShoreWare servers:

Total number of servers (i.e. sum of servers at all sites).
Number of servers at each site.
The name and IP address of every server.

The ShoreTel system also supports distributed voice application servers. Distributed servers help accomplish the following:

Reduce bandwidth, because local users' calls to voice mail are answered by the local voice mail application and do not go across the WAN.

Increase system scale by extending the unified messaging and desktop call control services to additional users of the applications.

Increase reliability by providing local support for some services and applications if a site loses connectivity with the Headquarters server.

Even though there are multiple servers, the ShoreTel system provides a single image system across your entire network. The system is currently certified to support up to 21 servers, one at the headquarters site and up to 20 distributed servers. You should add a server at any site that exceeds 100 users. You must deploy a server for every 1,000 users.

The distributed voice applications platform can also provide an open applications platform for extending telephone services through TAPI-compliant third-party applications. A dedicated distributed server is required to host the third-party applications. This server is deployed like other distributed servers, except that it must not have voice mail users assigned to it.

The distributed voice application servers' Remote TAPI Service Provider relies on the call control information from the main server. To add reliability to your remote server, consider using redundant network paths to the main server.

3.3.3 Citrix and Windows Terminal Server

Citrix and Windows Terminal Server (WTS) technologies enable processing for multiple users to be aggregated on a single Windows computer. The single Windows computer is a process- and disk-sharing server for multiple users who have lightweight or thin graphics stations on their desktop. Citrix communicates between the server and clients using the ICA protocol, whereas Windows Terminal Server uses the RDP protocol.

For more information on configuring ShoreTel Communicator clients on Citrix and WTS servers, see Appendix E, starting on page 277.

3.3.4 Teleworker Sites

In addition to the main locations, you can also deploy ShorePhone IP phones at employees' homes for the purpose of telecommuting. This allows teleworkers complete access to all the voice services on the network. The number and location of each teleworker IP phone should be noted on the topology diagram.

For information on configuring ShoreTel IP phones as teleworkers, see Chapter 16, starting on page 225.

3.3.5 Telephone Requirements

The next task in the system design process is to determine your telephone requirements.

To determine your telephone requirements:

Step 1 Count the telephones that are needed by counting the users installed on your current system. Make sure to include conference room telephones, lobby telephones, and telephones shared by multiple users.

Step 2 Determine the number of button boxes (ShoreTel BB24 devices) that will be needed for operators and receptionists. The maximum number is 4 BB24 devices per multiline phone.

Step 3 Determine the number of ports for fax machines and modems.

Step 4 If you are deploying IP phones, determine the number of telephones that will be IP phones and the number that will be analog phones.

Certain users will require access to certain features, such as an operator needing a phone with programmable buttons. Therefore, you should consider which type of functions each user will need in order to select the most appropriate phone for that user.

See Chapter 2, starting on page 23, for information on ShorePhone telephone types.

Step 5 Consider your needs for additional telephone ports for third-party systems, including conference bridges and overhead paging systems.

See Chapter 8, starting on page 101, for more information about selecting telephones.

Step 6 Determine the number of user licenses you need.

Each user on the system requires a user access license. The types of user licenses are listed below:

Extension and mailbox: Purchase of this license entitles the user to be assigned to both a physical extension and a ShoreTel voice mailbox.

Extension-only: Purchase of this license entitles the user to be assigned to a physical extension, either via explicit assignment or via Extension Assignment.

Mailbox-only: Purchase of this license allows entitles the user to be assigned to a ShoreTel voice mailbox.

An Extension-only user license is required for each conference room telephone, lobby telephone, fax machine, and modem user. Each port on a ShoreTel Conference Bridge also requires a user license (included with the bridge), however, a user access license is not required for trunks and anonymous telephones.

For more information about user licenses, see Chapter 18, starting on page 239.

Step 7 Fill in the telephone section of the Telephone and Trunk Planning Spreadsheet (Microsoft Excel Spreadsheet), shown in Figure 3-1.

The spreadsheet is available on the ShoreTel support web site for you to use in determining your telephone and trunk requirements. You must have Microsoft Excel to use this tool. If you are planning a multisite implementation, complete a telephone and trunk analysis for each site.

Telephones	**Quanity**	**IP Phone Use**	**Analog Ports**	**IP Phones**	**T1 Channels**	**E1 Channels**	**Server Capacity**
User Telephones	1,000	10%	900	100	-	-	1,000
Conference Room Telephones	20	10%	18	2	-	-	20
Lobby Telephones	2	10%	2	0	-	-	2
Modems	5	10%	5	1	-	-	5
FAX Machines	10	10%	9	1	-	-	10
Other	15	10%	14	2	-	-	15
Anonymous Telephones	20	10%	18	2	-	-	-
Virtual Users	50		-		-	-	50
Mailbox-only Users	75		-		-	-	-
Extension-only Users	50		-		-	-	-
Extension and mailbox Users	75		-		-	-	-
	----------		----------	----------	----------	----------	----------
Sub-Total	1,322		965	107	0	0	1,102
Trunks							
Analog Trunks	10		10		-	-	-
T1 Trunks (24 Channels / Span)	2		-		48	-	-
T1 PRI Trunks (23 Channels / Span)	4		-		92	-	-
E1 PRI Trunks (30 Channels / Span)	0		-		-	0	-
			----------		----------	----------	----------
Sub-Total			10		140	0	0
Total Trunks	150				**Trunks / Physical Telephones**		
Total Physical Telephones	1,072				**14%**		
ShoreGear Voice Switches							
ShoreGear-24	42						
ShoreGear-T1	6						
ShoreGear-E1	0						

Figure 3-1 Telephone and Trunk Planning Spreadsheet (Microsoft Excel)

3.3.6 Trunk Requirements

Trunks provide connectivity between users on the ShoreTel system and the public switched telephone network (PSTN). In this next task in the system design process, you determine the number of trunks required.

The number of trunks required on your system varies, depending on the number of users and your specific application needs. It is important to size your trunking correctly because not having enough trunks can lead to blocked calls when all trunks are busy, and too many trunks can lead to wasted money on monthly access charges.

See Chapter 5, starting on page 67, for more information about trunk features, ordering, and installation.

You have several options for determining the number of trunks your site requires:

- Review the number of trunks on your current system. In general, this is one of the best methods to gauge the number of trunks you need.
- You can also request a traffic analysis from your service provider, interconnect, or telecom manager to understand your current trunk utilization. This method will help you understand your current usage and allow you to maintain the current service level.
- Visit a web site, such as www.erlang.com, to use a traffic calculator for determining your trunk requirements.
- Fill in the Trunks section of the spreadsheet shown in Figure 3-1 to determine the number of trunks you need. The spreadsheet automatically calculates the trunking ratio.
- Consider Table 3-2 and the following:
  - Larger locations can typically use lower-density trunking (15%).
  - Smaller locations need higher-density trunking (50%).
  - Some applications, such as call centers, can demand higher-density trunking (50%).

Trunk Density	Trunks/Users %
Low	15%
Average	30%
High	50%

Table 3-2 Trunk Density

When planning trunks, consider the call volume for your workgroups or ACD groups. Since there is generally a queuing solution in place for ACD calls, the number of trunks required should be based on the full utilization of the expected number of agents and sufficient trunks for the expected number of waiting callers.

3.4 Determine Number of ShoreGear Voice Switches

The ShoreTel system is a software solution that runs on standard platforms across the network equipment in your enterprise. The ShoreGear hardware portfolio offers a broad family of voice switches to meet the needs of our different customers. Each ShoreGear voice switch connects to your IP network using a 10/100M auto-sensing Ethernet port. Refer to Appendix G, starting on page 303 for a description of all ShoreGear switches.

To determine the number of voice switches:

Fill in the ShoreGear voice switches section in the Telephone and Trunk Planning Spreadsheet (Figure 3-1) to calculate the number of voice switches required.

When you compute the user and trunk information in the spreadsheet, the number of switches for each site is provided.

See Appendix A, starting on page 263, for more information about which voice switches and features are supported in countries other than the United States.

3.4.1 WAN Connections

To complete your system design, the final step is to identify your network connectivity. You should identify the following for the network connections to each site:

- Bandwidth
- Latency
- Jitter
- Packet Loss

CHAPTER 4

Routing Calls

This chapter helps you identify the desired routing for inbound and outbound calls, so that you can determine your requirements for configuration and trunking.

When installing a voice communications system, one of the most important decisions you must make is how to route incoming calls. This includes calls made to your company, an individual employee, or a group of employees, such as sales or customer support. It is important to consider not only how calls are initially routed, but also how they are routed when the person or group is not available to take the call. Will calls be transferred to the Auto-Attendant, the operator, an off-site number, a pager, or a cell phone? The ShoreTel system is highly flexible and supports numerous methods to route incoming calls.

Task	Description
Direct All Calls to an Auto-Attendant	page 56
Direct All Calls to a Live Operator	page 58
Direct All Calls to Extensions	page 60
Blended Call Routing	page 62
Analyze Outbound Call Routing	page 64

Table 4-1 Routing CallsChecklist

In addition, you must consider your outbound call routing plan. You should have trunks at every site that supports both outbound and inbound calling.

This chapter helps you design the call flow of your new voice communications system. See Chapter 3, starting on page 49, for information about other aspects of designing your new voice communications system.

If you are installing a ShoreTel Contact Center Solution, call routing within the contact center is configured separately and is not covered in this guide. For more information on the ShoreTel Contact Center Solution, see the *ShoreTel Contact Center Solution Administration Guide*.

4.1 Recommendations

Consider the following recommendations when designing your call flow plan:

- Determine how calls should reach employees and workgroups. You need to identify the desired call routing for inbound calls at each site.
- Identify contingencies, such as alternate plans in the event that the receptionist has an unplanned absence, or the physical phone fails. For example, creating hunt groups can ensure an operator is available if the receptionist or workgroup is unavailable.

Consider the inter-site call flow, such as your operator's or receptionist's role in handling inbound calls, and the role of others who are not physically present at the main site.

Identify call flow early. Do not wait until the last minute, or try to identify the call flow the day of cut-over.

Interview the key members of your organization (workgroups, operators, assistants, and executives) to determine their individual preferences and needs, and make sure they agree with any decisions that affect their respective areas.

Create an off-hours call routing plan.

4.2 Hunt Groups

Hunt groups allow you to route calls to a list of extensions. Hunt groups can be accessed through an extension, DID, and/or DNIS. Hunt groups are supported by ShoreGear switches and remain available even when connectivity to the Headquarters server is lost. A single switch can host up to 8 hunt groups and a maximum of 16 extensions total per switch. A hunt group can be used as the backup destination for an operator or workgroup, so that basic hunting occurs even when the operator or workgroup is not reachable. To maximize reliability, assign hunt groups to a switch close to the majority of the members and/or trunks associated with the hunt group.

Hunt groups can be used for:

Backup Routing for a workgroup

Hunt groups can be used when the workgroup server is not reachable because of a network outage or admission control. When the hunt group is set to offer each member a single call at a time, then call offering is similar to a workgroup.

Hunt Group as a Call Forward Destination

In a small office where individuals generally receive calls directly, users may want someone in the office to answer calls when they are unable to answer. Hunt groups can provide alternate destinations in this case.

Distribution of Calls to Backup Operators

A hunt group can provide backup operators for the primary operator who handles calls to a main company number.

Common Line Monitoring

A hunt group can be used for line monitoring. For example, several operators may wish to monitor the same line and all have an opportunity to answer calls at the same time.

4.3 Direct All Calls to an Auto-Attendant

You can direct all inbound calls to the automated attendant, and prompt the calling party to route the call, based on menu options. Auto-attendant answering is typically used by smaller companies and smaller locations that do not choose to use direct inward dial (DID) numbers. See Figure 4-1 for an illustration of auto-attendant call flow.

Organize the auto-attendant with options for various departments. In addition, include an "out" for callers if they must speak to a live attendant or have a rotary telephone. This destination must be one that will always be answered. In many cases, it is a receptionist's extension that is staffed at all times, or a night chime that can be answered by any employee. If you route calls to a receptionist's position that is not always staffed or the receptionist needs to be mobile, consider installing a cordless telephone for the receptionist to wear while roaming around the office. If this is not an option, make sure the receptionist's call handling modes are set up appropriately.

4.3.1 Trunk Considerations

An auto-attendant menu can be reached through analog loop-start, digital loop-start, and T1/E1 PRI trunks by pointing the trunk group at the desired menu. You can also reach a specific menu using DID or DNIS entries received over analog wink-start, digital wink-start, or T1/E1 PRI trunks.

The ShoreTel system supports International Caller ID, Caller ID Name, Caller ID Number, ANI, and DNIS. The Caller ID and trunk group or DNIS information is provided to the user to assist in answering the call.

4.3.1.1 Call Routing and Collecting Caller ID Information

The switch delays each inbound loop-start call by 1.5 rings to collect caller ID information before ringing the user's telephone. This allows caller ID information to reach the user's client at the time the call rings the extension, rather than after it rings the extension.

Features available on trunks vary by trunk type. See Chapter 5, starting on page 67, for more information.

4.3.2 After-Hours Call Routing

For after hours, weekends, and holidays, consider how your call flow will change. Typically, a different prompt is played, since callers are routed directly to voice mail rather than to workgroups or the operator.

4.3.3 Example of Auto-Attendant Call Routing

In the call flow example shown in Figure 4-1, all calls are received by the auto-attendant. The calling party can choose to be directed to:

The support workgroup by dialing a digit.

Calls are presented to the support workgroup with a mailbox that provides coverage. The calling party can dial “0” in the mailbox to reach the workgroup assistant, or “9” to return to the auto-attendant.

An employee using Dial by Number or Dial by Name.

Calls are presented to the employee with a mailbox that provides coverage. The calling party can dial “0” in the mailbox to reach the employee's personal assistant, or “9” to return to the auto-attendant.

The operator by dialing the digit 0.

Calls are presented to the operator. If the operator does not answer, a backup operator provides coverage using the operator's call handling modes. If the backup operator does not answer, a mailbox provides coverage, and the calling party can dial “0” in the mailbox to reach the operator's personal assistant, or “9” to return to the auto-attendant.

In this example, the workgroup, users, and operator route calls directly to voice mail after hours.

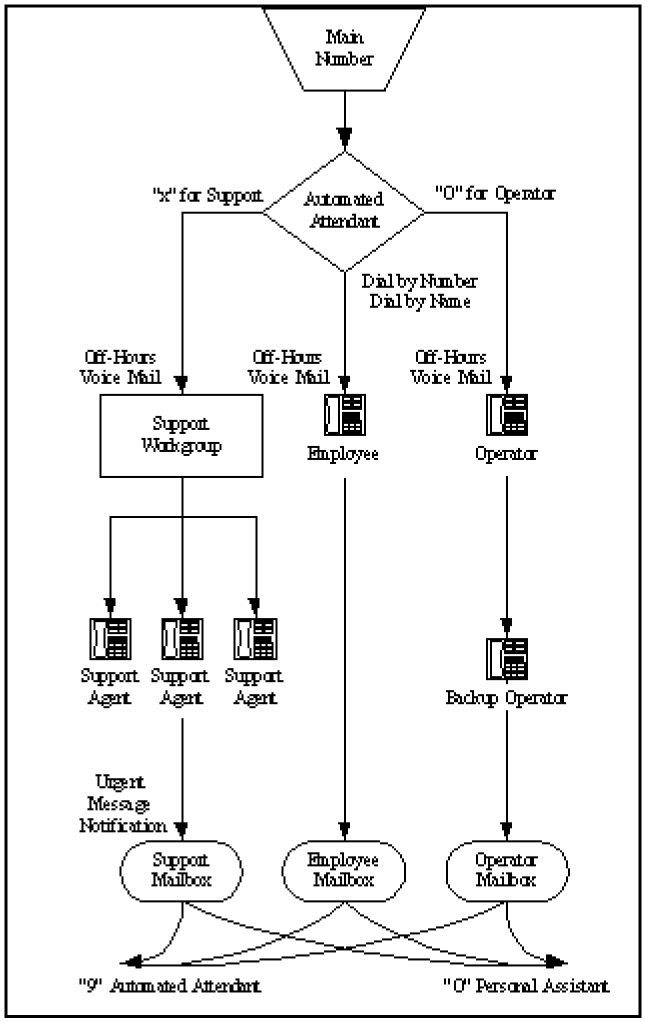

Figure 4-1 Auto-Attendant Call Routing

4.4 Direct All Calls to a Live Operator

Some companies choose to answer all inbound calls during business hours with a live operator to give callers a more personal experience. If you use a live operator, the most important thing to remember is that the operator's telephone must always be staffed. ShoreTel recommends the following:

Use the ShoreTel Communicator - Operator Access software, because the standard telephone without ShoreTel Communicator manages only a single call at a time. When a second call arrives, using the Flash button invokes call waiting, generating a swap hold situation in which calls cannot be transferred. This problem is eliminated when you use the ShoreTel Communicator - Operator Access software.

If the organization is a large one, consider using the ShorePhone-BB24 button box. The button box provides additional shortcut functions for ShorePhone multiline phones. The button box behaves like an additional set of 24 custom buttons that can be used by the operator to quickly and easily route calls to executives and to other employees who receive a high volume of phone calls. A maximum of 4 BB24 devices can be connected to each multiline phone.

If the operator does not receive a lot of telephone calls and is required to roam around the office to deliver mail, pick up faxes, make copies, and so on, a two-line 2.4 GHz cordless telephone can be used. The first line is reserved for incoming calls, while the second line is the operator's personal extension.

Create hunt groups to ensure someone is always available to take an incoming call.

You can choose to have calls initially routed to the operator and then forwarded to the auto-attendant after a fixed number of rings.

Operators work in either of two modes:

Answer all calls and transfer them to the appropriate destination.
Answer all calls and hold them until the parties are found.

If your operator works in the second mode, you should consider installing an overhead paging system or should consider using the Paging Groups feature (see the ***ShoreTel Administration Guide*** for details on Paging Groups).

The ShoreTel system supports single-zone overhead paging on a per-site basis, using the audio output jack on the switches supplied with the jack. When you need multiple-zone paging, please use ShoreTel's online knowledge base, to access the application note on paging on ShoreTel's web site at www.shoretel.com.

4.4.1 Trunk Considerations

The operator can be reached through analog loop-start, digital loop-start, and T1/E1 PRI trunks by pointing the trunk group directly at the operator. You can also reach the operator using DID or DNIS entries received over analog wink-start, digital wink-start, or T1/E1 PRI trunks.

The ShoreTel system supports International Caller ID, Caller ID Name, Caller ID Number, ANI, and DNIS. The Caller ID and trunk group or DNIS information is provided to the user to assist in answering the call.

Features available on trunks vary by trunk type. See Chapter 5, starting on page 67, for more information.

4.4.2 After-Hours Call Routing

If you route all calls to the operator's extension, auto-attendant scheduling does not apply; only those calls routed to the auto-attendant use the schedule. Therefore, if you want to use the off-hours, holiday, and custom schedules, set the operator's call handling mode to forward all calls to the auto-attendant when the operator is unavailable.

4.4.3 Example Using Hunt Groups

To route calls to a prioritized list of backup operators, create hunt groups with users who can serve as backup operators. In this scenario, a primary operator who handles calls to a main company number requires one or more secondary operators to receive the calls when the primary operator becomes too busy.

To create a hunt group to back up the primary operator:

Create a hunt group with backup operators.

Enter the main operator and all the backups as members of the hunt group in the order in which they are to serve as backups.

Set the hunt group for multiple calls to be hunted to a given member.

Set the call stack size for each of the users to control the number of calls he or she can receive.

When there are incoming calls to the hunt group, the primary operator is offered the calls first. The operator may be offered multiple calls concurrently up to the limit of his or her call stack. If a member's call stack is full, the member is skipped and that particular call is not be offered again (unless the hunt group is set to hunt forever and no member picks up the call before the member is reached again in the hunt list).

If a member of the operator group does not answer the hunt call, the call is offered to the next member after the number of configured rings. Thus, even if the primary operator has room on his or her call stack, the call is offered to the next member in the list when the operator does not answer the call in time.

For more information on Hunt Groups, see Section 12.10 on page 171.

4.4.4 Example of Operator Call Routing

In the example call flow shown in Figure 4-2, all calls are received by the operator, who then transfers the calls to the appropriate destination.

Calls are transferred to the support workgroup with a mailbox that provides coverage.

The calling party can dial "0" in the mailbox to reach the workgroup assistant, or "9" to return to the auto-attendant.

Calls are transferred to the employees with a mailbox that provides coverage.

The calling party can dial "0" in the mailbox to reach his or her personal assistant, or "9" to return to the auto-attendant.

If the operator does not answer, a backup operator provides coverage, using the operator's call handling modes.

If the backup operator does not answer, a mailbox provides coverage and the calling party can dial "0" in the mailbox to reach the operator's personal assistant, or "9" to return to the auto-attendant.

In this example, after-hours call routing is handled by an auto-attendant in a very similar fashion to the previous example (Figure 4-1). To start after-hours call handling, the operator changes his or her call handling mode. This can be done automatically using Microsoft Outlook Calendar in conjunction with Automated Call Handling (although it does require the operator's personal computer to remain connected with Microsoft Outlook running on it).

4.5 Direct All Calls to Extensions

ShoreTel recommends using Direct Inward Dial (DID) trunks so that callers can dial extensions directly without having to go through the operator. This provides the most efficient, professional call handling experience to your customers.

In the event that an individual is not available, preconfigured call handling modes route callers. This routing might include a cellular telephone, a pager, an alternate extension, or a personal assistant. Additionally, consider using the voice mail notification capabilities of the ShoreTel system when employees are not able to answer the telephone but need to stay in touch.

Even if you choose to direct all calls to extensions, you should still configure the auto-attendant for Dial by Number, Dial by Name, and zero out to an operator.

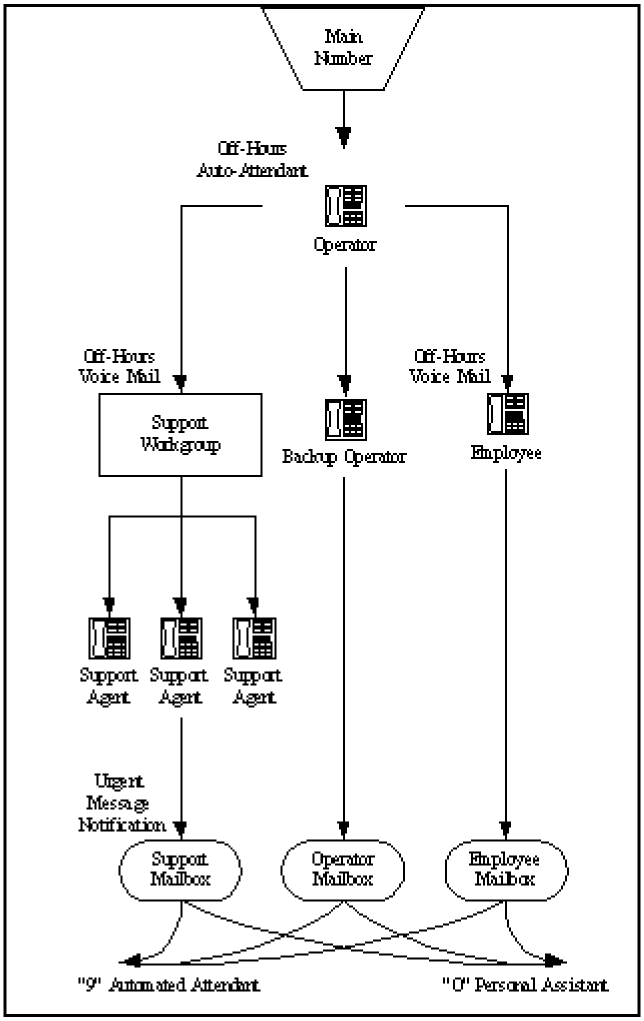

Figure 4-2 Operator Call Routing

4.5.1 Trunk Considerations

When using Direct Inward Dial, you must use analog wink-start, digital wink-start, SIP or T1/E1 PRI trunks. The ShoreTel system can receive Automatic Number Identification (ANI) over analog and digital wink-start trunks as well as Caller ID Number over T1/E1 PRI.

Features available on trunks vary by trunk type. See Chapter 5, starting on page 67, for more information.

4.5.2 After-Hours Call Routing

By routing all calls to the individual extensions, each individual user and workgroup defines its after-hours call handling.

4.5.3 Example of Direct Inward Dial Call Routing

In the illustration shown in Figure 4-3, all calls are received by workgroups or by individuals.

Calls are routed directly to the support workgroup with a mailbox that provides coverage.

The calling party can dial “0” in the mailbox to reach the workgroup assistant or “9” to return to the auto-attendant.

Calls are routed directly to the employees with a mailbox that provides coverage.

The calling party can dial “0” in the mailbox to reach his or her personal assistant, or “9” to return to the auto-attendant.

An operator provides limited call handling functions from individual mailboxes or the automated attendant.

In this example, after-hours call routing is received by the workgroups and individual employees.

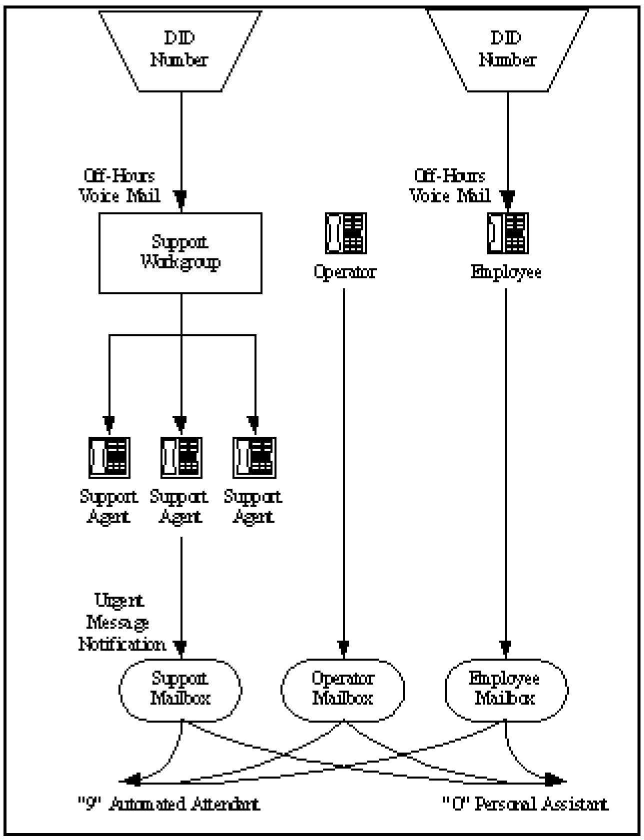

Figure 4-3 Direct Inward Dial Call Routing

4.6 Blended Call Routing

Communication systems typically use a mix of automated, live, and DID call routing to maximize user satisfaction as well as efficiency and flexibility. This usually includes taking a published main telephone number and routing it to the auto-attendant, as well as installing DID lines that route calls directly to different workgroups and individual employees.

4.6.1 Trunk Considerations

An auto-attendant menu can be reached through analog loop-start, digital loop-start, SIP, and T1/E1 PRI trunks by pointing the trunk group at the desired menu. You can also reach a specific menu using DID or DNIS entries received over analog wink-start, digital wink-start, or T1/E1 PRI trunks.

The operator can be reached through analog loop-start, digital loop-start, and T1/E1 PRI trunks by pointing the trunk group directly at the operator. You can also reach the operator using DID or DNIS entries received over analog wink-start, digital wink-start, or T1/E1 PRI trunks.

The ShoreTel system supports International Caller ID, Caller ID Name, Caller ID Number, ANI, and DNIS. The Caller ID and trunk group or DNIS information will be provided to the user to assist in answering the call.

When using Direct Inward Dial, you must use analog wink-start, digital-wink start, or T1/E1 PRI trunks. The ShoreTel system can receive Automatic Number Identification (ANI) over analog and digital wink-start trunks as well as Caller ID Number over T1/E1 PRI.

Features available on trunks vary by trunk type. See Chapter 5, starting on page 67, for more information.

4.6.2 After-Hours Call Routing

For after hours, weekends, and holidays, you should consider how your call flow will change. Typically, a different prompt should be played, since callers are routed directly to voice mail rather than to workgroups or the operator.

If you route all calls to the operator's extension, auto-attendant scheduling does not apply; only those calls routed to the auto-attendant use the schedule. Therefore, when you want to use the off-hours, holiday, and custom schedules, set the operator's call handling mode to forward all calls to the auto-attendant when unavailable.

By routing all calls to the individual extensions, each individual user and workgroup defines its after-hours call handling.

4.6.3 Example of Blended Call Routing

In the example shown in Figure 4-4, a mix of inbound call routing is used.

- Calls are routed directly to the support workgroup using DID and DNIS entries and routed through the auto-attendant with a mailbox that provides coverage.
- The calling party can dial “0” in the mailbox to reach the workgroup assistant, or “9” to return to the auto-attendant.
- Calls are routed directly to the employees using DID and routed through the auto-attendant using Dial by Number and Dial by Name with a mailbox that provides coverage.
- The calling party can dial “0” in the mailbox to reach his or her personal assistant, or “9” to return to the auto-attendant.
- An operator provides limited call handling functions from individual mailboxes or the auto-attendant.

In this example, after-hours call routing changes at the auto-attendant and for each of the workgroups, employees, and the operator, because each workgroup defines its own after-hours call routing.

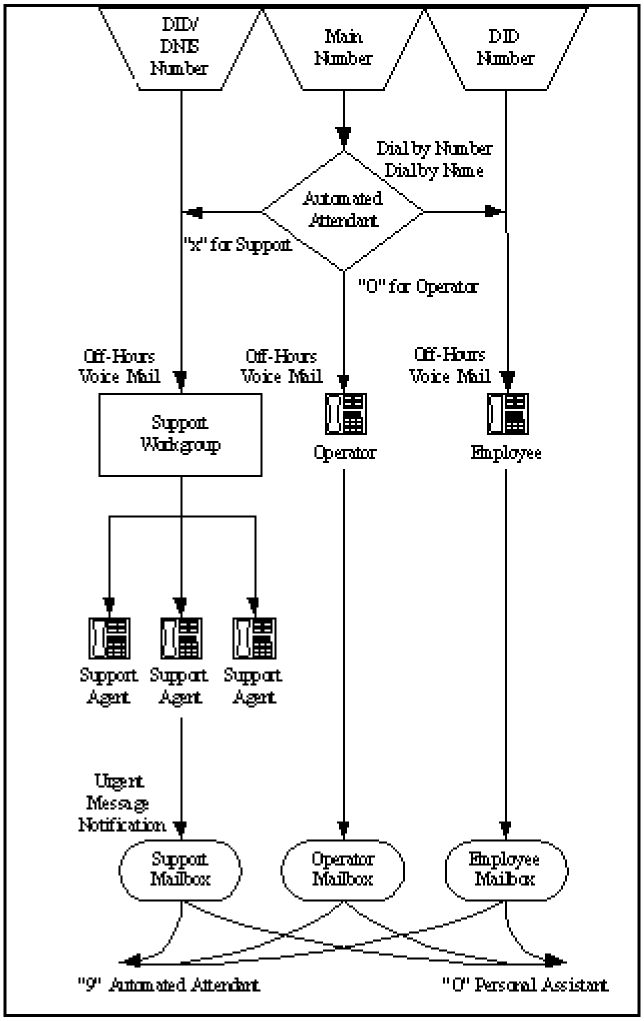

Figure 4-4 Blended Call Routing

4.7 Analyze Outbound Call Routing

In general, you should have trunks at every site that support both outbound and inbound calling. Here are some general comments about outbound trunking:

ISDN PRI provides the most feature-rich inbound and outbound calling experience.

This includes the support for Caller ID, DID, and DNIS. Caller ID Number is supported for both inbound and outbound calls. Caller ID Name is supported only on inbound NI-2 trunks (with the exception of outbound calls to off-system extensions).

SIP trunks can be used to place outbound calls.

Analog wink-start trunks do not support outbound calls.

You may want to purchase some analog loop-start trunks for emergency dial tone in case of total power failure. For more information, see Section 5.3.1 on page 68.

Calls can be automatically routed across your wide area network (WAN) using the Network Call Routing feature. (This allows users to access local and “nearby” area codes at one site from another site.)

You need to plan for emergency calls (such as 911 in the United States) on your voice system.

The ShoreTel system supports all the necessary signaling for emergency calls. Please see the appendix on emergency 911 operations in the ***ShoreTel Administration Guide*** for information on how to configure your system for emergency calls.

If your system uses three-digit extensions, ShoreTel recommends that you do not assign x11 extensions to users.

For more information, see Chapter 5, starting on page 67, and Chapter 6, starting on page 81.

CHAPTER 5

Trunk Planning and Ordering

This chapter explains the features and functionality of trunks on the ShoreTel system, so you can plan and order your service. It includes the following information:

- An overview of the trunk types supported on the ShoreTel system
- A description of each trunk feature
- Traffic calculations
- Trunk ordering and order form

5.1 Checklist

You must complete the following tasks before proceeding to the next chapter:

Task	Description
Reviewing and Selecting Trunk Types	page 67
Understanding Trunk Features	page 72
Performing Traffic Calculations	page 76
Ordering Telephone Service	page 76

Table 5-1 Trunk Planning and Ordering Checklist

5.2 Recommendations

The following recommendations assist you in determining your trunk requirements and ordering your trunks from your service provider:

- Make sure you order telephone service early. T1 and PRI service can take up to one or two months to install.
- If you are reusing Centrex lines, be sure to change your old service and remove call waiting, call forwarding, and voice mail.
- When provisioning PRI service, be sure to confirm the protocol being used (NI-2, 4ESS, 5ESS, or DMS-100). Make sure that neither NFAS nor the Call-by-Call feature of the 4ESS is being used, since they are not supported on the ShoreTel system.

5.3 Reviewing and Selecting Trunk Types

Trunks provide a connection from the ShoreTel system to a service provider for the purpose of making and taking calls to and from external parties.

Table 5-2 shows which trunk types are supported on individual ShoreGear switches. The next section provides more detailed information about the various trunk features.

Voice Switch	Analog Loop-Start (N.Am.)	Analog Loop-Start (EMEA)	Digital Loop-Start	Analog Wink-Start	Digital Wink-Start	T1 PRI	E1 PRI	SIP	BRI
ShoreGear 90	Yes	Yes	No	Yes	No	No	No	Yes	No
ShoreGear 90BRI	No	No	No	No	No	No	No	Yes	Yes
ShoreGear 50	Yes	Yes	No	Yes	No	No	No	Yes	No
ShoreGear 30	Yes	Yes	No	Yes	No	No	No	Yes	No
ShoreGear 220E1	No	No	No	No	No	No	Yes	Yes	No
ShoreGear 220T1	No	No	Yes	No	Yes	Yes	No	Yes	No
ShoreGear 220T1A	Yes	No	Yes	No	Yes	Yes	No	Yes	No
ShoreGear T1k	No	No	Yes	No	Yes	Yes	No	No	No
ShoreGear 120	Yes	No	No	Yes	No	No	No	Yes	No
ShoreGear 60	Yes	No	No	Yes	No	No	No	Yes	No
ShoreGear 40	Yes	No	No	Yes	No	No	No	Yes	No
ShoreGear E1	No	No	No	No	No	No	Yes	No	No
ShoreGear T1	No	No	Yes	No	Yes	Yes	No	No	No

Table 5-2 Supported Trunk Types

5.3.1 Analog Loop-Start Trunks (North America)

Analog loop-start trunks are typically used for inbound calls to a main telephone number that are directed to an auto-attendant menu, company operator, or workgroup. A caller can route a call from the auto-attendant to a user extension by entering the extension number or by spelling the user's name from the telephone keypad. Analog loop-start trunks are also used to make outbound calls.

Analog loop-start trunks support:

- Inbound calls
- Outbound calls
- Caller ID number
- Caller ID name
- Caller ID blocking

Analog provisioning is provided by the loop-start protocol and Dual-Tone Multi-Frequency (DTMF) signaling.

Analog loop-start trunks are used to provide power-fail transfer to selected telephones—for instance, to the operator, security station, executives, and so on. When there is a complete power failure, including loss of UPS power backup, power-fail transfer connects a specified trunk port to a specified extension port. This power-fail transfer ability provides a dial tone for making and taking critical calls in the event of power failure. Refer to Appendix G, starting on page 303, for the power-fail transfer port on each ShoreGear switch that supports this feature.

Centrex lines are analog lines that can be used as analog loop-start trunks. Your organization may already have these installed, and want to use them instead of ordering new loop-start trunks. If you have Centrex lines and do not want to change your primary company telephone number, you can keep Centrex lines. Centrex lines support Caller ID. Be sure to remove the Centrex features, including call waiting, call forward, and voice mail.

EMEA analog loop start trunk support, based on the TBR 21 standard, is supported on all 1U Half Width ShoreWare voice switches. BT type 1 (on hook) caller ID support is based on SIN 227 and SIN 242 standards in the UK.

5.3.2 Analog Loop-Start Trunks (EMEA)

Analog Loop-Start trunks are supported in Europe, the Middle East, and Africa, and are based on the TBR 21 standard.

Analog Loop-Start Trunks (EMEA) are typically used for inbound calls to a main telephone number that are directed to an auto-attendant menu, company operator, or workgroup. A caller can route a call from the auto-attendant to a user extension by entering the extension number or by spelling the user's name from the telephone keypad. Analog loop-start trunks are also used to make outbound calls.

Analog loop-start trunks (EMEA) support:

- Inbound calls
- Outbound calls
- BT type 1 (on hook) caller ID support is based on SIN 227 and SIN 242 standards in the UK.

Analog provisioning is provided by the loop-start protocol and Dual-Tone Multi-Frequency (DTMF) signaling.

Analog loop-start trunks are used to provide power-fail transfer to selected telephones—for instance, to the operator, security station, executives, and so on. When there is a complete power failure, including loss of UPS power backup, the ShoreGear switches provides power-fail transfer. Refer to Appendix G, starting on page 303, for the power-fail transfer port on each ShoreGear switch that supports this feature. This power-fail transfer ability provides a dial tone for making and taking critical calls in the event of power failure.

Centrex lines are analog lines that can be used as analog loop-start trunks on the ShoreGear switches. Your organization may already have these installed, and want to use them instead of ordering new loop-start trunks. If you have Centrex lines and do not want to change your primary company telephone number, you can keep Centrex lines. Centrex lines support Caller ID. Be sure to remove the Centrex features, including call waiting, call forward, and voice mail.

5.3.3 Digital Loop-Start Trunks

Digital loop-start trunks are typically used for inbound calls to the main telephone number that are directed to an auto-attendant menu, company operator, or workgroup. A caller can route a call from the auto-attendant to a user extension by entering the extension number or by spelling the user's name from the telephone keypad. Digital loop-start trunks are also used to make outbound calls.

Digital loop-start trunks support:

- Inbound calls
- Outbound calls
- Caller ID number
- Caller ID name
- Caller ID blocking

Digital provisioning is provided by the loop-start protocol and Dual-Tone Multi-Frequency (DTMF) signaling. ShoreGear switches support

ESF or D4 framing formats
B8ZS or AMI line coding.

5.3.4 Analog Wink-Start Trunks (Analog DID)

Analog wink-start trunks allow external callers to dial a user's phone number directly, without having to use an auto-attendant or operator. Analog wink-start trunks support only inbound calls; they are not capable of handling outbound calls.

Analog wink-start trunks support:

Inbound calls (outbound calls are not supported)
ANI
DID
DNIS

Analog provisioning is provided by the wink-start protocol and Dual-Tone Multi-Frequency (DTMF) signaling.

If ANI is being used, the star (*) key must be used to delimit the ANI digits from the DID/DNIS digits—that is:

<DID>
<DNIS>
<ANI><DID/DNIS>*

5.3.5 Digital Wink-Start Trunks

Digital wink-start trunks allow external callers to dial a user's phone number directly, without having to use an auto-attendant or operator. Digital wink-start trunks support both inbound and outbound calls.

Digital wink-start trunks support:

Inbound calls
Outbound calls
ANI
DID
DNIS

Digital provisioning is provided by the wink-start protocol (often called E&M wink-start) and Dual-Tone Multi-Frequency (DTMF) signaling. ShoreGear switches support

ESF or D4 framing formats
B8ZS or AMI line coding.

If ANI is being used, the star (*) key must be used to delimit the ANI digits from the DID/DNIS digits—that is:

<DID>
<DNIS>
<ANI><DID/DNIS>*

5.3.6 BRI Trunks

BRI trunks are flexible trunks that support both inbound and outbound calls.

PRI trunks support:

Inbound calls
Outbound calls
DID
DNIS
Caller ID number
Caller ID name is supported for NI-2 configured trunks
QSIG – Calling name is supported if the standard is similar to NI2
Inbound calling name is fully supported, but outbound calling name is only supported for Off-System Extension calls

Digital provisioning is provided by the PRI protocol and D-channel signaling. ShoreGear switches support

DMS-100, 4ESS, 5ESS, and NI-2 signaling types
ESF or D4 framing formats
B8ZS or AMI line coding.

The NFAS and Call-by-Call features are not supported.

5.3.7 T1 PRI Trunks

T1 PRI trunks are flexible trunks that support both inbound and outbound calls.

PRI trunks support:

Inbound calls
Outbound calls
DID
DNIS
Caller ID number
Caller ID name is supported for NI-2 configured trunks
QSIG – Calling name is supported if the standard is similar to NI2
Inbound calling name is fully supported, but outbound calling name is only supported for Off-System Extension calls

Digital provisioning is provided by the PRI protocol and D-channel signaling. ShoreGear switch supports

DMS-100, 4ESS, 5ESS, and NI-2 signaling types
ESF or D4 framing formats
B8ZS or AMI line coding.

The NFAS and Call-by-Call features are not supported.

5.3.8 E1 PRI Trunks

E1 PRI trunks are flexible trunks that support both inbound and outbound calls for international locations.

E1 PRI trunks support:

Inbound calls
Outbound calls
DID
DNIS
Caller ID number
Caller ID name is supported for NI-2 configured trunks

QSIG – Calling name is supported if the standard is similar to NI2
Inbound calling name is fully supported, but outbound calling name is only supported for Off-System Extension calls

The ShoreGear switches support PRI signaling using Euro-ISDN as well as other international protocols. See Appendix A, starting on page 263.

5.3.9 SIP Trunks

SIP trunks are flexible trunks that support both inbound and outbound calls. SIP trunks are logical trunk end points that only handle SIP call control. Media flows directly between the call initiator and the call terminator.

SIP trunks support:

Inbound calls
Outbound calls
Extension, Tandem, and default destinations for inbound calls
Caller ID name
Caller ID number
DID
DNIS

By default, the "Enable SIP Info for G711 DTMF signaling" check box is off. This check box must be enabled for ShoreTel-to-ShoreTel SIP tie trunks or for SIP devices that do not support RFC 2833 for G711.

5.4 Understanding Trunk Features

The ShoreTel system supports several different trunk types and trunk features. It is very important to understand the features available on these trunks, since some services are mutually exclusive. Table 5-3 shows each trunk type and the associated features

Feature	Analog Loop-Start N.Am.	Analog Loop-Start EMEA	Digital Loop-Start	Analog Wink-Start	Digital Wink-Start	T1 PRI	E1 PRI	SIP	BRI
Inbound:									
• Caller ID Number	Yes	No	Yes	Yes[a]	Yes[a]	Yes	Yes	Yes	Yes
• Caller ID Name	Yes	No	Yes	No	No	Yes[b]	Yes	Yes	Yes
• Direct Inward Dial (DID)	No	No	No	Yes	Yes	Yes	Yes	Yes	Yes
• Dialed Number Identification Service (DNIS)	No	No	No	Yes	Yes	Yes	Yes	Yes	Yes
Outbound:									
• Caller ID Blocked	Yes (CO)	Yes (CO)	Yes (CO)	N/A	Yes (CO)	Yes	Yes	Yes	Yes
• Caller ID Unblocked	Yes (CO)	Yes (CO)	Yes (CO)	N/A	Yes (CO)	Yes	Yes	Yes	Yes
• Caller ID Blocking Override (*67, *82)	Yes[c]	No	Yes[c]	N/A	No	Yes	No	No	No

Table 5-3 Trunk Features

a. Via Automatic Number Identification (ANI).

b. Caller ID Name is supported for NI-2 configured trunks.
c. *67 and *82 codes do not work if the CO requires a pause between the code and the dialed number.

Legend to Table 5-3

Yes—Feature is supported.
No—Feature is not supported.
Yes (CO)—Feature is provided by the central office (CO) or legacy PBX.
N/A—Outbound calls are not supported on analog wink-start trunks.

5.4.1 Caller ID Number

Caller ID Number delivers to the ShoreTel system the number of the calling party, which is displayed in the ShoreTel Communicator as well as on Caller ID–compatible telephones. The delivery of the caller ID number can be blocked by the calling party. The caller ID number is delivered unless the calling party has blocked the call (in which case the call is marked as “Blocked”), or the service provider does not have the information (in which case the call is marked as “Unavailable”).

Caller ID Number has the following limitations:

The calling party may block his or her caller ID number.

The calling party may be calling from a business and the calling number may be incorrect.

The calling party may be calling from someone else’s number.

Caller ID Number is available on analog loop-start, digital loop-start, SIP, T1 PRI, and E1 PRI trunks.

Two different Caller ID Number formats are used to deliver caller information via loop-start trunks: Single Data Message Format (SDMF) and Multiple Data Message Format (MDMF). SDMF provides the calling number, while MDMF provides any combination of calling name and number. The ShoreGear voice switches support both SMDF and MDMF dynamically, without the need for configuration. When PRI is used, the caller ID number is delivered as a D-Channel message.

ShoreTel supports International Caller ID, ensuring that when a switch is configured for a certain site (e.g. Spain), the International ID information is automatically filled in as appropriate for that country. The feature is transparent from the user's standpoint, and no configuration is necessary.

5.4.2 Caller ID Name

Caller ID Name delivers the name of the calling party to the ShoreTel system. The name is displayed in the ShoreTel Communicator as well as on any telephones that support caller ID Name.

By default, the caller ID name is delivered unless the calling party has blocked the transfer of this information (in which case the call is marked as “Blocked”). If the service provider does not have the information, the call is marked as “Unavailable.”

Caller ID Name is available on analog loop-start and digital loop-start trunks, as well as SIP, T1 PRI, and E1 PRI trunks and is only supported on IP phone and analog phones in North America. This feature is not supported on analog phones in other countries.

When using NI-2 signaling on PRI trunks—for example in a tie-trunk scenario—Caller-ID Name is now also captured when available on all inbound calls. For outbound calls, Caller-ID Name is delivered for calls that are made to off-system extensions, but not for outbound calls.

5.4.3 Automatic Number Identification (ANI)

Automatic Number Identification (ANI) delivers the number of the calling party to the ShoreTel system. Although similar to Caller ID Number, ANI is tariffed differently and is not subject to the same blocking restrictions as Caller ID Number. For instance, when you purchase ANI services from your service provider, you are always delivered the calling number for 800-number calls (calls that you are paying for). This may vary from region to region.

ANI is available on analog wink-start and digital wink-start trunks.

When ANI is being used, the star key (*) must be used to delimit the ANI digits from the DID/DNIS digits—that is, *<ANI>*<DID/DNIS>*.

5.4.4 Direct Inward Dial (DID)

Direct Inward Dial (DID) allows extensions (users, menus, workgroups, route points, etc.) on the system to be accessed directly, without the need of an auto-attendant or operator. This is particularly useful when users on the system want their own telephone number.

DID is available on analog wink-start, digital wink-start, PRI and SIP trunks.

DID numbers are ordered in blocks of 20 or more 10-digit telephone numbers. These numbers are assigned to a customer and are routed to a wink-start, PRI or SIP trunk connected to a voice switch. When a call is made, the service provider sends a predefined set of digits (from 3 to 10 digits) via the wink-start, PRI, or SIP trunk. The voice switches capture the digits and route the calling party to the called party.

If ANI is not being used on wink-start trunks, only the DNIS digits need to be delivered. If ANI is being used, the star (*) key must be used to delimit the ANI digits from the DID/DNIS digits—that is:

<DID>
<DNIS>
<ANI><DID/DNIS>*

5.4.5 Dialed Number Identification Service (DNIS)

Dialed Number Identification Service (DNIS) allows extensions (users, menus, workgroups, route points, etc.) on the system to be accessed directly, without the need of an auto-attendant or operator. This is particularly useful for workgroup and other call center applications. The DNIS information is delivered to the ShoreTel Communicator - Personal Access and stored in the call detail record.

DNIS is available on analog wink-start, digital wink-start, PRI and SIP trunks.

DNIS numbers are ordered individually and map to a dialed number. When a calling party dials a specific telephone number, the service provider routes the call to a wink-start or PRI trunk connected to a voice switch. The service provider sends a predefined set of digits (from 3 to 10 digits)—the DNIS digits—using DTMF signaling (or a D-Channel message or SIP message). The voice switches capture the digits and route the calling party to the called party.

If ANI is not being used on wink-start trunks, only the DNIS digits need to be delivered. If ANI is being used, the star (*) key must be used to delimit the ANI digits from the DID/DNIS digits—that is:

<DID>
<DNIS>
<ANI><DID/DNIS>*

5.4.6 Outbound Caller ID

ShoreTel sends the user's DID number as the caller ID number for outbound calls over PRI or SIP trunks. If the DID number is unavailable, the site Caller Emergency Service ID (CESID) is used. If that number is unavailable, no caller ID is sent.

Additionally, the outbound caller ID can be configured on a per-user basis such that the configured value can take precedence over the user's DID number or the site CESID. Note that this feature is only available on outbound calls using a T1 PRI trunk.

To send a single main number rather than individual user DID numbers, assign DNIS entries instead of DID numbers to each user. The Site Contact Number will be sent on outbound calls.

To block all outbound caller ID numbers from being sent, you can configure the PRI trunk group to always block the caller ID number.

On wink-start and loop-start trunks, the outbound caller ID is defined by the service provider.

On T1 PRI and loop-start trunks, users can override the Caller ID Blocking configuration on a call-by-call basis by using commands at the telephone (*67, *82). Users cannot override the Caller ID Blocking configuration of wink-start and E1 PRI trunks.

For more information on configuring outbound caller ID, please refer to the ***ShoreTel Administration Guide***.

5.4.7 Tandem Trunking

Tandem trunking allows legacy voice systems to utilize a ShoreTel system for outbound dialing. The ShoreTel system supports both user-side and network-side PRI, allowing ShoreTel systems to flexibly support digital tie trunks to other systems.

You can enable tandem trunking support for any PRI trunk group with a check box in ShoreWare Director. Tandem calls are associated with a user group for outbound trunk selection. Inbound calls recognized as tandem calls are redirected to an outbound trunk based on user group call permissions and trunk group access. When needed, a "dial-in prefix" can be specified that is prepended to digits collected on tandem calls. The concatenated set of digits is then used in outbound trunk selection for the tandem call.

5.4.8 Tie Trunks

The addition of network-side PRI support makes PRI tie trunks easier and more compelling to deploy. ShoreGear switches that support T1 PRI can act as either the user-side or network-side of a PRI tie trunk. The tie trunk may be used to tie a ShoreTel system to a legacy voice system, or potentially to another independent ShoreTel system.

5.5 Performing Traffic Calculations

The number of trunks required on your system will vary depending on the number of users and your specific application needs. It is important to order your trunking correctly; too few can lead to blocked calls when all trunks are busy, and too many trunks can lead to wasted money on monthly access charges.

See Chapter 3, starting on page 49, for information about calculating the trunk requirements for your site.

5.6 Ordering Telephone Service

Once you have determined the types of trunks you need, you will have to either place a new order or make a change order. You can use the associated "Telephone Service Order Forms" that are available on the ShoreWare DVD or on the ShoreTel support web site. Three order forms are provided for your use:

- Analog Service
- T1 Service
- T1 PRI Service

ShoreTel does not provide an E1 PRI form because this service varies by country. Instead, we provide a table of the E1 PRI parameters that must be set. See Appendix A, starting on page 263, for more information.

When the form is completed, arrange a meeting with your telephone company service representative to order the new telephone services. The forms contain specific information that the service representative must have before services can be ordered.

Before ordering your telephone service, pay special attention to the installation date and time, as follows:

- If you are **ordering new service**, it should be installed one week before the planned cut-over date. This allows the services to be terminated on the ShoreTel system and tested before cut-over.

- If you are **changing existing service**, any changes before the cut-over date might render your existing service unusable. You must schedule these changes outside normal business hours and work closely with your service provider for a seamless transition.

When ordering DID service, the last digits of the DID numbers should match your extension numbers for ease of use. You must make sure your extension numbers do not begin with a trunk access code, zero, or any emergency numbers such as 911 in North America.

Please see the appendix on emergency 911 operations in the ***ShoreTel Administration Guide*** for information on how to configure your system for emergency calls.

5.6.1 Analog Service

Use the Analog Telephone Service Order form (Figure 5-1) to order analog trunks. Note the following about analog service:

- Caller ID Name and Number are supported on loop-start trunks.
- ANI is supported on wink-start trunks.
- ANI on wink-start trunks must be delivered as *<ANI>*<DNIS>*.

Telephone Service Order - Analog Trunks		
Customer Name:		
Today's Date:		
Cut-over Date:		
Cut-over Time:		
Vendor		
Make:	ShoreTel, Inc.	
Model:	ShoreGear	
FCC Registration Number:	4ABUSA-26003-MF-E	
Ringer Equivalence:	0.5B	
Analog Loop Start Trunks		
Quantity of Trunks:		
Protocol:	Loop Start	
Signalling:	DTMF	
Caller ID:	Yes / No	
Caller ID Format:	MDMF	
Caller ID Delivery:	Blocked / Unblocked	
Analog Wink Trunks		
Quantity of Trunks:		
Protocol:	Wink Start	
Signalling:	DTMF	
Automatic Number Identification (ANI):	Yes / No	
Direct Inward Dial (DID):	Yes / No	
Quanity of Numbers (Block)		Example: 100, 200
Number of Digits:	3 - 4 - 5 - 6 - 7 - 8 - 9 - 10	
Dialed Number Identication Service (DNIS):	Yes / No	
Quanity of Numbers:		
Number of Digits:	3 - 4 - 5 - 6 - 7 - 8 - 9 - 10	
Comments		

Figure 5-1 Telephone Service Order Form—Analog Trunks

5.6.2 T1 Service

Use the T1 Telephone Service Order form (Figure 5-2) to order T1 trunks. Note the following about T1 service:

Caller ID Name and Number are supported on loop-start trunks.
ANI is supported on wink-start trunks.
ANI on wink-start trunks must be delivered as *<ANI>*<DNIS>*.
A channel service unit (CSU) is built into the voice switch.

5.6.3 T1 PRI Service

Use the T1 PRI Telephone Service Order form (Figure 5-3) to order T1 PRI trunks. Note the following about T1 PRI service:

Caller ID Number is supported on T1 PRI trunks. (Caller ID Name is supported in NI-2 configured trunks.)

A channel service unit (CSU) is built into the voice switch.

5.6.4 Ordering Service

When you order service, be sure to do the following:

Telephone Service Order - T1 Trunks		
Customer Name:		
Today's Date:		
Cut-over Date:		
Cut-over Time:		
Vendor		
Make:	ShoreTel, Inc.	
Model:	ShoreGear	
FCC Registration Number:	4ABUSA-26003-MF-E	
Ringer Equivalence:	0.5B	
Digital Loop Start Trunks		
Quantity of Trunks:		
Protocol:	Loop Start	
Signalling:	DTMF	
Framing Format:	ESF / D4	
Line Code:	B8ZS / AMI	
Caller ID:	Yes / No	
Caller ID Format:	MDMF	
Caller ID Delivery:	Blocked / Unblocked	
Digital Wink Trunks		
Quantity of Trunks:		
Protocol:	Wink Start	
Signalling:	DTMF	
Framing Format:	ESF / D4	
Line Code:	B8ZS / AMI	
Automatic Number Identification (ANI):	Yes / No	
Direct Inward Dial (DID):	Yes / No	
Quanity of Numbers (Block)		Example: 100, 200
Number of Digits:	3 - 4 - 5 - 6 - 7 - 8 - 9 - 10	
Dialed Number Identication Service (DNIS):	Yes / No	
Quanity of Numbers:		
Number of Digits:	3 - 4 - 5 - 6 - 7 - 8 - 9 - 10	
Comments		

Figure 5-2 **Telephone Service Order Form—T1 Trunks**

State that a new ShoreTel system is being installed.

State the date and time the new telephone service must be cut over.

Review all the items on the telephone service order form with the service representative.

Review any existing and new telephone numbers and have the telephone company representative confirm the order.

5.6.5 E1 PRI Service

See Appendix A, starting on page 263, for more information about ordering E1 PRI service.

Telephone Service Order - PRI Trunks

Customer Name:		
Today's Date:		
Cut-over Date:		
Cut-over Time:		
Vendor		
Make:	ShoreTel, Inc.	
Model:	ShoreGear	
FCC Registration Number:	4ABUSA-26003-MF-E	
Ringer Equivalence:	0.5B	
PRI Trunks		
Quantity of Trunks:		
Protocol:	PRI	
Central Office Type:	4ESS / 5ESS / DMS-100 / NI-2	
Signalling:	DTMF	
Framing Format:	ESF / D4	
Line Code:	B8ZS / AMI	
Service:	Inbound / Outbound / Both	
Caller ID:	Yes / No	
Caller ID Delivery:	Blocked / Unblocked	
Direct Inward Dial (DID):	Yes / No	
Quanity of Numbers (Block)		Example: 100, 200
Number of Digits:	3 - 4 - 5 - 6 - 7 - 8 - 9 - 10	
Dialed Number Identication Service (DNIS):	Yes / No	
Quanity of Numbers:		
Number of Digits:	3 - 4 - 5 - 6 - 7 - 8 - 9 - 10	
Comments		

Figure 5-3 Telephone Server Order Form—PRI Trunks

CHAPTER 6

Dialing Plan

This chapter provides an overview of the dialing, call routing, and digit-manipulation capabilities of the ShoreTel system.

The information in this chapter is particularly useful for administrators of larger, multisite installations.

6.1 Overview

When a phone number is dialed in a ShoreTel system, the system performs two distinct operations on a telephone number:

Digit collection. Voice switches collect the digits in a telephone number.

Digit manipulation. The switches manipulate the dialed numbers before outpulsing them to the service provider.

In this chapter you will learn how to define what happens at each of these steps. Once you are familiar with these concepts, we will introduce you to On-Net Dialing, a feature that allows users to divide phone numbers into two separately-managed parts for a more flexible dialing plan.

6.2 Checklist

Before configuring your phones (but **after** mapping out your network and trunk configuration), you need to perform the tasks in the table below:

Task	Description
Define Digit Collection	page 81
Define Digit Manipulation	page 87
On-Net Dialing	page 88

Table 6-1 Dialing Plan Checklist

6.3 Define Digit Collection

When someone picks up a telephone in a ShoreTel system and begins dialing a telephone number, the voice switch software examines each digit in the number and determines whether digit collection should continue or be terminated.

6.3.1 Configuring Internal Numbers

In a ShoreTel system where users dial internal numbers without an access code, the rules for digit collection are relatively straightforward.

Digit collection rules are configured through ShoreWare Director. To view the *Dialing Plan* edit page, click *Dialing Plan* under *System Parameters*. Figure 6-1 shows the *Dialing Plan* edit page.

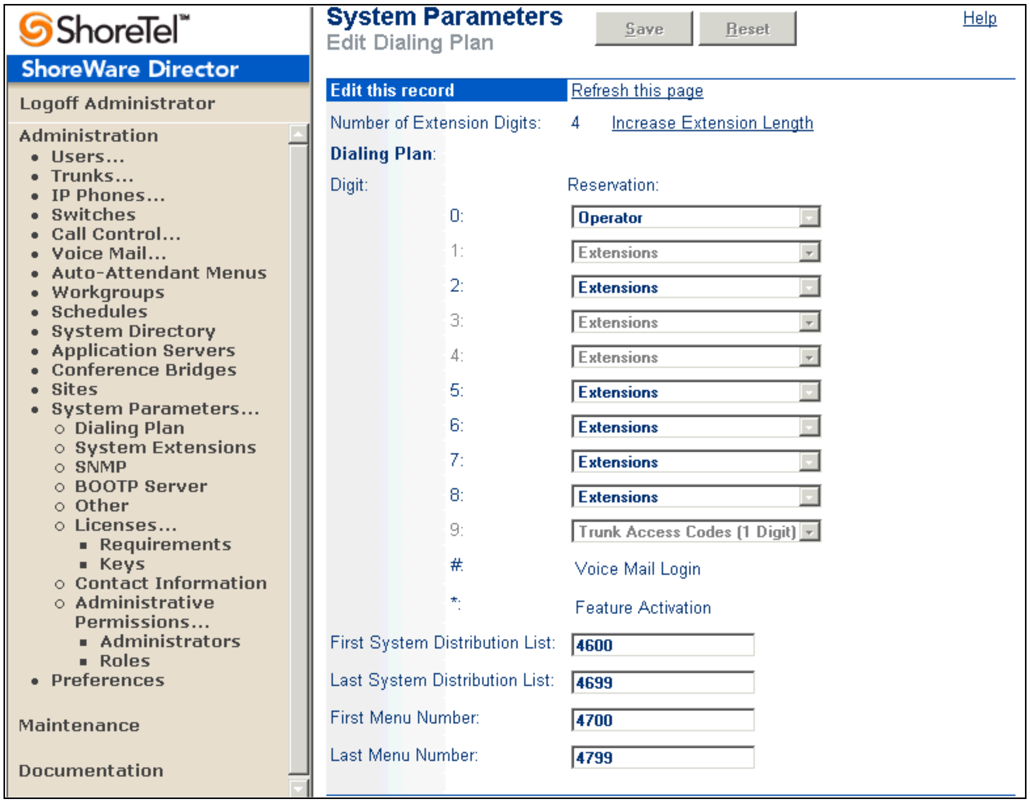

Figure 6-1 Dialing Plan Edit Page

6.3.1.1 Planning Your Dialing Configuration

When setting up a dialing plan for internal numbers, you need to consider the following:

Choose an extension length. ShoreTel supports 3-, 4-, and 5-digit dialing for internal numbers (4-digit dialing works for most enterprises). Use an extension number scheme that conforms to your company's size and the convenience of your users.

Map extension ranges. After choosing the extension length, you can allocate blocks of numbers for use by extension, starting with the first number.

For example, if you want to reserve the range of numbers 3000-3999 for extension assignment, you allocate the "3" number block for extensions.

For maximum usability, map extension numbers to the final digits of your DID (if DID is used).

Extensions cannot begin with "911" (911, 911x, or 911xx).

6.3.1.2 Digit Collection Rules

When routing calls, the ShoreTel system follows the digit collection rules specified on the *Dialing Plan* edit page in ShoreWare Director.

For the first digit collected, specific rules are in effect.

Digit	Rule
0 – The digit configured in the dialing plan as the operator digit.	Digit collection is stopped and the call is routed to the site operator.
#	Digit collection is stopped and the call is routed to voice mail login.
Any other digit	Digit collection continues until a complete extension number is dialed. If the number is valid, the call is routed to the extension. • valid off-system extensions – the call is routed to a trunk. • invalid extensions – the call is routed to the Backup Automated Attendant. Rule does not apply to trunk access codes.

Table 6-2 Digit Collection Rules

6.3.1.3 Exception for 911 Emergency Calls

Emergency calls do not require an access code.

The following rules apply only to emergency 911 calls:

If "911" is dialed, the switch routes the call to a 911-capable trunk group associated with the caller's User Group.

Before switching the emergency call, the switch invokes a brief timeout for insurance against accidental 911 calls. If any digit is entered during the timeout, the switch routes the call to the Backup Automated Attendant.

Although this section focuses on emergency calls made within the United States, the same rules apply in other countries. See the appendix on emergency 911 operations in the ***ShoreTel Administration Guide*** for information on how to configure your system for emergency calls.

To define digit collection for internal numbers:

In ShoreWare Director, go to the ***Dialing Plan*** edit page under ***System Parameters*** and edit the dialing plan parameters. See the ***ShoreTel Administration Guide*** for a description of the parameters on this page.

6.3.1.4 Changing Extension Length

The ShoreTel system supports 3-, 4-, and 5-digit extensions.

To change the extension length:

Step 1 Click ***Increase Extension Length***.

Step 2 Specify 4 or 5 digits for the increased length.

After applying your edits, you cannot decrease an extension length. For example, once it is increased to 4, the minimum is 4.

If your system uses three-digit extensions, ShoreTel recommends that you do not assign x11 extensions to users.

6.3.2 Configuring External Numbers

The ShoreTel system supports 1-, 2-, and 3-digit trunk access codes. When an access code is dialed, the system looks for a valid digit in the parameters.

If an invalid number is dialed, the system plays a recording to the calling party.

There are several types of valid telephone numbers, which are described in the following sections.

The ShoreTel system allows the system administrator to provide users at each site with a unique dialing plan to match the dialing plan of the site's geographic region. The ShoreTel system supports 7-digit local dialing, 10-digit local dialing, and mixed local dialing.

External numbers are converted into a standard "canonical format" by call control software to provide a globally consistent way of handling phone numbers. The canonical format starts with a "+" representing the international prefix, followed by the country code, area code, and subscriber number.

> External numbers that can be converted into canonical format are considered "routable" and will leverage the network call routing feature of the call control software.
>
> External numbers that are unique to the country (n11, 112, 911, and so on) are considered "unroutable" and will not leverage the network call routing software. These calls will be placed from the local site or the associated proxy site.

6.3.2.1 Configuring 7-Digit Local Dialing

The ***Local Area Code*** on the ***Site*** edit page, shown in Figure 6-2, defines 7-digit dialing for all users at the site. When a user dials an access code followed by 7 digits, the switching software assumes the site local area has been dialed. The switching software then converts the 7-digit number into canonical format before checking call permissions and doing network call routing.

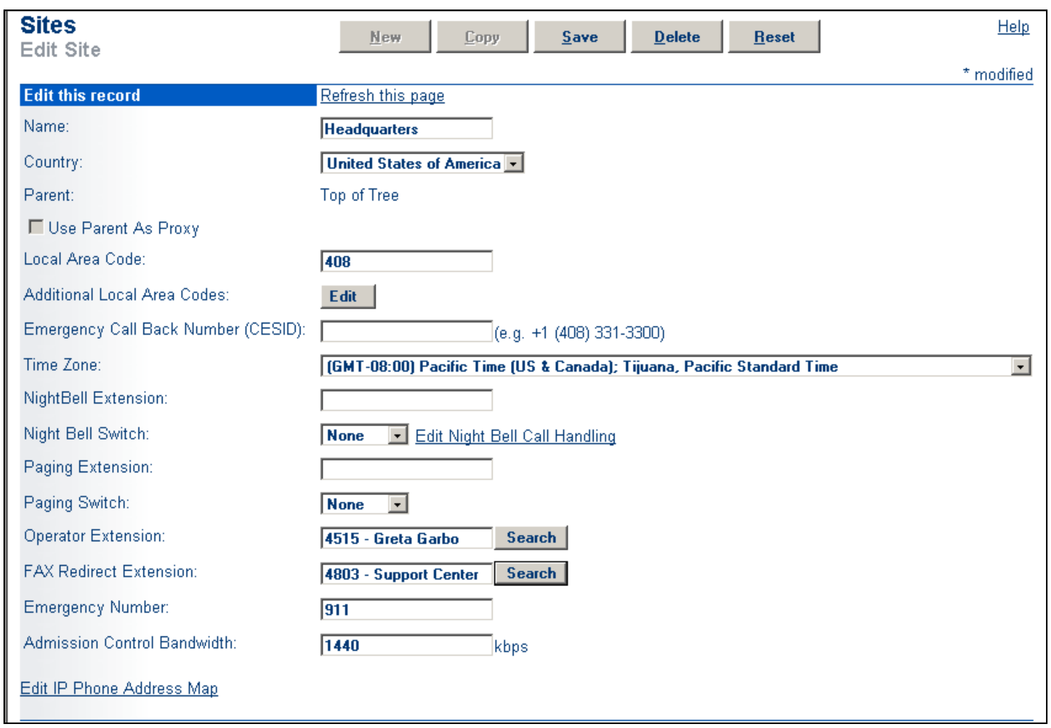

Figure 6-2 Site Edit Page

The ***Local Area Code*** and ***Additional Local Area Codes*** set on the ***Site*** edit page have nothing to do with the ***Local Area Code***, ***Additional Local Area Codes***, and ***Nearby Area Codes*** on the ***Trunk Group*** edit page. Area codes on the ***Site*** edit page relate only to digit collection, whereas those on the ***Trunk Group*** edit page relate only to Network Call Routing and Digit Manipulation.

To define 7-digit dialing:

Step 1 Open the ***Site*** edit page in ShoreWare Director.

Step 2 Enter the 3-digit area code in the ***Local Area Code*** field.

See the ***ShoreTel Administration Guide*** for more information about the ***Site*** edit page.

6.3.2.2 Configuring 10-Digit Local Dialing

If the site is in a location with overlay area codes, it can be configured to support 10-digit dialing for all the local area codes. The ***Additional Local Area Codes*** field on the **Site** edit page defines the area codes for 10-digit dialing. When a user dials an access code followed by a local area code, the system collects 7 additional digits (10 digits total) before stopping digit collection. The switching software then converts the 10-digit number into canonical format before checking call permissions and doing network call routing.

To define 10-digit dialing:

Step 1 Open the ***Site*** edit page in ShoreWare Director.

Step 2 Click ***Edit*** next to the ***Additional Local Area Codes*** field.

The ***Additional Local Area Codes*** dialog box, shown in Figure 6-3, appears.

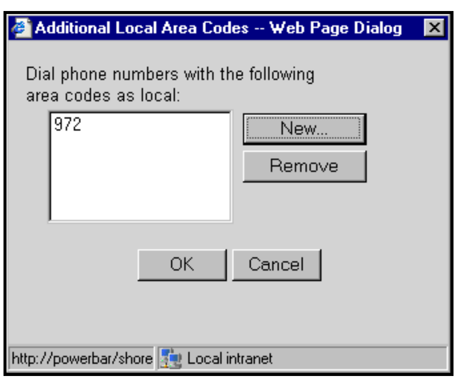

Figure 6-3 Additional Local Area Codes Dialog Box

See the ***ShoreTel Administration Guide*** for more information about the ***Additional Local Area Codes*** field on the ***Site*** edit page.

6.3.2.3 Configuring Mixed Dialing in the Same Area

In locations where users are forced to dial 7 digits for some prefixes and 1+10 digits for other prefixes in the same area, the ShoreTel system supports permissive dialing—that is, you can dial these numbers either as 7 digits or as 1+10 digits. It also supports permissive dialing in locations with mixed 10-digit and 1+10 digit dialing in the same area.

From a digit-manipulation (or outpulsing) point of view, the trunk group must be configured properly since some service providers do not support permissive dialing. See Section 6.4 on page 87.

6.3.2.4 1+10 Digit Long-Distance Dialing

The ShoreTel system supports long-distance dialing. When a user dials an access code followed by "1," the software collects 10 additional digits before stopping digit collection.

6.3.2.5 International Dialing

The ShoreTel system supports international dialing. If the user dials a trunk access code followed by an international access code, digit collection is terminated after a timeout. The timeout can be bypassed by dialing pound (#).

6.3.2.6 n11 Dialing

The ShoreTel system supports "n11" dialing, including 411 (information) and 611 (support). If the user dials an access code followed by "n11," digit collection is terminated after a brief timeout and the call is routed to a trunk.

If your system uses three-digit extensions, ShoreTel recommends that you do not assign x11 extensions to users.

6.3.2.7 911 Dialing

The ShoreTel system supports 911 dialing to emergency services. If the user dials an access code followed by "911," digit collection is terminated immediately and the call is routed to a trunk.

911 calls are routed out of the local site's associated trunks. If there are no 911 trunks available at the local site, the call is routed via the designated proxy site.

Please see the appendix on emergency 911 operations in the ***ShoreTel Administration Guide*** for information on how to configure your system for emergency calls.

6.3.2.8 Explicit Carrier Selection (101xxxx) Dialing

The ShoreTel system supports explicit carrier selection. If the user dials an access code followed by "101," the next four digits collected are for explicit carrier selection (101xxxx). The carrier information is retained and passed to the trunk.The digits collected are treated as unroutable calls; the digits are routed "as-is" out either local site or proxy site trunks only.

6.3.2.9 Operator-Assisted (0, 00) Dialing

The ShoreTel system supports operator-assisted dialing. If the user dials an access code followed by "0x," digit collection is terminated after a brief timeout and the call is routed to a trunk.

6.3.2.10 Vertical Service Code (*67, *82) Dialing

The ShoreTel system supports some vertical service codes for feature activation. If the user dials an access code followed by star (*), subsequent digits are collected and terminated by a brief timeout. The digits collected are treated as unroutable calls—they are routed "as-is" out either local site or proxy site trunks only. If the trunk used is a PRI trunk, that trunk strips and interprets *67 to block outbound Caller ID, and *82 to unblock outbound Caller ID.

6.3.2.11 End Digit Collection (#)

In some cases, digit collection ends after a timeout period. To bypass the timeout and route the call immediately, dial pound (#).

6.4 Define Digit Manipulation

Once the route decision has been made, the call is passed to the trunk. The dialed number, which is normally passed within the system in canonical format, is examined and manipulated based on the trunk group configuration. This ensures that the number can be properly received by the service provider.

First, the trunk access code dialed by the user is removed. If the number is in canonical format (local, long distance, ERC, international), digit manipulation can occur. If the number is unroutable (n11, ECS, operator, and vertical service code numbers) digit manipulation (other than the dial-out prefix) is not applied.

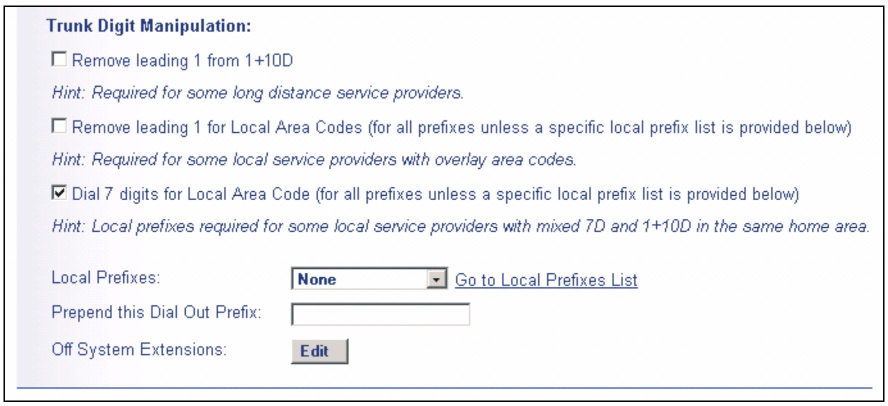

Figure 6-4 Digit Manipulation on the Trunk Group Edit Page

To specify trunk digit manipulation:

Step 1 Open the ***Trunk Digit Manipulation*** page, shown in Figure 6-4.

Step 2 Select the options and specify numbers as needed, using Table 6-3 as a guide.

Option	Description	Example
Remove leading 1 from 1+10D	This option is required by some long-distance service providers that only accept numbers dialed as 10 digits.	AT&T typically only supports 10-digit dialing.
Remove leading 1 for Local Area Codes	This option is required by some local service providers that have mixed 10-digit and 1+10 digit dialing in the same area code. Local Area Codes include both the Local Area Code and Additional Local Area Codes configured against the trunk group.	Atlanta has three local area codes that must be dialed as 10 digits. This could also be called “Dial 10 digits for Local Area Codes.”
Dial 7 digits for Local Area Code	This option is required by some local service providers that have mixed 10-digit and 1+10 digit dialing in the same area code.	Massachusetts and Maine.

Table 6-3 Digit Manipulation Options

<table>
<tr><th>Option</th><th>Description</th><th>Example</th></tr>
<tr><td>Prepend this Dial Out Prefix</td><td>The Dial Out Prefix is prepended to the number. This feature is typically used when connecting the ShoreTel system to a legacy PBX system using a ShoreGear voice switch. The Dial Out Prefix enables the ShoreTel system to seize a trunk on the legacy PBX. The Dial Out Prefix is not applied to Off-System Extensions.</td><td>Not applicable.</td></tr>
<tr><td>Vertical Service Codes</td><td>If a Vertical Service Code was dialed, digit manipulation rules do not apply.

Vertical Service Codes work with ISDN PRI trunks and some loop-start trunks.

• With PRI trunks, Vertical Service Codes for Caller ID Blocking control will be converted to D-Channel messages.
• With loop-start trunks, the service provider must be able to accept the outpulsed digits with only 50 msecs of pause between each digit, including the service codes.

Vertical Service Codes are typically not supported by service providers on wink-start trunks. If you have outbound access on wink-start trunks and you dial a vertical service code, you will likely get an error message from the service provider.</td><td>Not applicable.</td></tr>
<tr><td>Off System Extensions</td><td>Off System Extensions define ranges of extensions that when dialed will be routed out of this trunk group. This is typically used to interface to a legacy PBX system using a T1 or E1 circuit provided by a ShoreGear voice switch. Off-system extensions digits can be manipulated using a translation table.

Digit manipulation, including the Dial Out Prefix, will not be applied to these calls.</td><td>Not applicable.</td></tr>
</table>

Table 6-3 Digit Manipulation Options

6.5 On-Net Dialing

ShoreTel supports On-Net Dialing (OND), an enhancement that allows users to create more flexible dialing plans than before. In contrast with previous releases which could only support a “flat” dialing plan and treated all numbers as a single, indivisible unit, the On-Net Dialing feature allows users to divide phone numbers into two separately-managed parts:

extension prefix - typically 3 digits in length; similar in concept to a site code

user extension - typically 4 digits in length; acts as the number you would dial to reach other users in your organization

By dividing phone numbers into two parts, the OND feature provides customers with a more seamless method of migrating from their legacy phone systems to the newer ShoreTel system. OND allows customers to preserve their existing dialing plans when integrating ShoreTel equipment with their legacy equipment. While previous releases allowed customers to integrate ShoreTel equipment with their legacy PBX, the configurations needed to maintain the customer's existing dialing plan were complex and the complexity increased with the number of people and extensions involved.

For example, if one company acquired another company and the two companies wanted to merge their phone systems, then no two users could have the same user extension, even if they were at different sites with different prefixes.

With OND, users will be able to call other users within a site by dialing only the user extension. Inter-site calls would require users to dial the extension prefix plus the user number. Off-system extensions (OSE's) will continue to be used to route calls to legacy PBX's.

How It Works

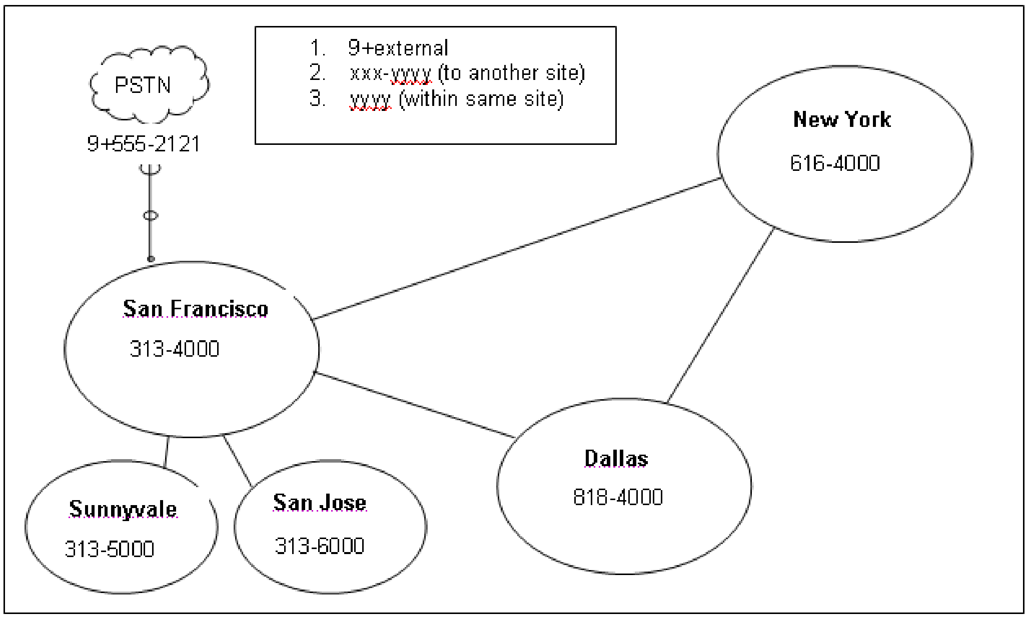

Figure 6-5 Abbreviated 4-digit dialing with extension prefix

As shown in the illustration above, On-Net Dialing assigns extension prefixes to each site or to a group of sites. All calls are placed “on the network” if they are within the same prefix, and the user need only dial the user extension. Calls preceded with the trunk access code (usually “9”) are sent to the PSTN.

Benefits of On-Net Dialing:

Scalability – For larger organizations, On-Net Dialing enables the creation of a common and consistent “cookie cutter” dialing plan that can be replicated throughout an organization that has many offices. For example, a department store might have a phone in each of its different departments with one for clothing, furniture, kitchenware, etc. With On-Net Dialing, a user can assign the extensions of 4000, 5000, 6000, and 7000 to each of these departments. By modifying the 3-digit site code/extension prefix at each location, this approach of assigning 4-digit extensions to departments can be replicated across an entire department store, nationwide, so that a user who knows the extension for the automotive department in one city could travel to another city and would know how to reach the automotive department if he knew the site code.

Preserve existing legacy dialing plans – As mentioned before, you can preserve the existing dialing plans when adding ShoreTel equipment to a deployment with legacy equipment by assigning a new prefix to each new site or to users on the new ShoreTel system.

Legacy integration via OSEs (Off-System Extensions) – Ability to call multiple legacy PBXs from the ShoreTel system.

Multi-tenant – On-Net Dialing allows a landlord to maintain one phone system at a building that houses two or more businesses or organizations in such a way that neither organization is aware that the infrastructure or trunk lines are being shared. Despite the fact that both organizations are in the same building, you can assign different prefixes to each company and could then hide one organization's phone numbers from the other group so that neither group would see the other via the directory.

6.5.1 Configuration

The process of configuring On-Net Dialing consists of the following tasks:

Planning and Configuring the Dialing Plan
Adding Sites
Associating an Extension Prefix with a Site
Assigning User Extensions

Each of these tasks is addressed in more detail below.

Enabling On-Net Dialing is an irreversible process that makes permanent changes to the database. Thus, you should plan carefully before proceeding with any configuration changes.

6.5.1.1 Planning and Configuring the Dialing Plan

Assigning extension prefixes to a specific digit must be done all at once. Once the dialing plan window (shown below) has been configured and saved, there is no way to make changes to the extension prefix assignments without erasing the database and starting all over. Therefore, we recommend carefully planning and reviewing your dialing plans before configuring the dialing plan window.

To configure the dialing plan via Director, follow the procedure below:

Step 1 Launch **ShoreWare 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 **System Parameters** link and then the **Dialing Plan** link to display the Edit Dialing Plan window, as shown below:

Step 4 Click on the drop-down menu to the right of the desired digit and select the number of digits you would like the extension prefix (i.e. site code) to contain. Extension prefixes can range from 1 to 7 digits in length. The leading digit determines the length of the prefix. Extension prefixes with different leading digits do not have to contain the same number of digits.

Step 5 Repeat this process for any other extension prefixes, unused extensions, or trunk access codes.

Step 6 Click **Save** to store your changes. The **Configure Extension Prefix Warning** window (similar to the one shown below) appears with a list of each of the sites in your system.

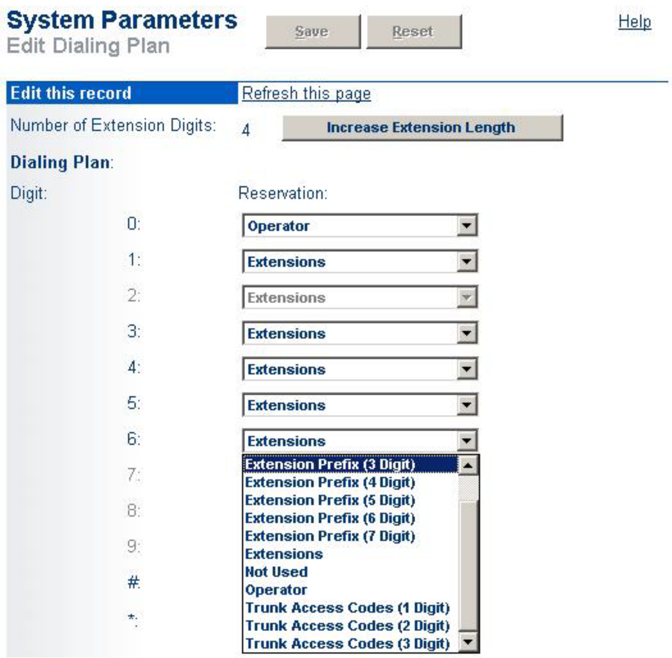

Figure 6-6 Configuring dialing plan

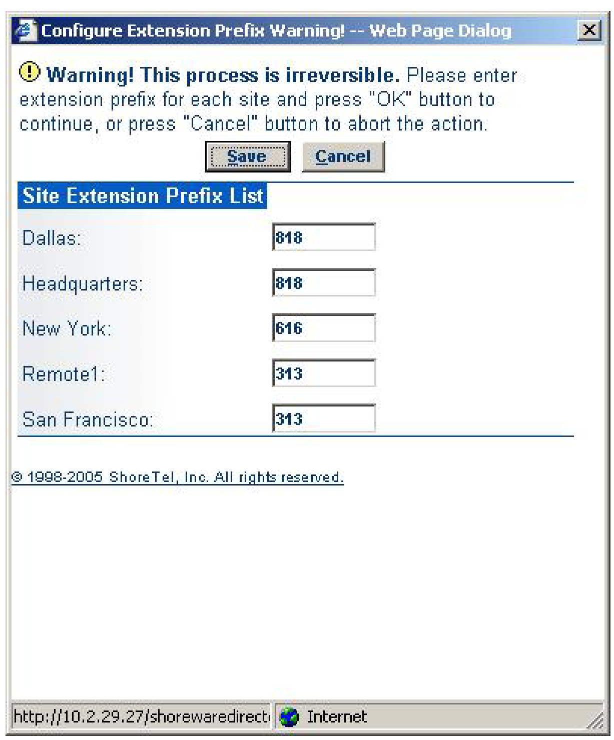

Figure 6-7 Make sure to back up your system before clicking Save

The Extension Prefix Warning message lists each site in your system. Next to the list of sites you will find a blank field that requires you to enter the desired extension prefix. Note that this prefix will be applied to every dialable number at that particular site, so if the site is an existing one, they will see their phone numbers converted to the new prefix.

System extensions are not associated with a hard port in the system. They are always global and will have a user number and a null extension prefix. Therefore, these system extensions are not affected by changes made to the extension prefix in the Edit Dialing Plan window. Only dialed numbers (user extensions, menus, workgroups, distribution lists) are affected by changes to the extension prefix.

Step 7 Click the **Save** button to store your changes.

6.5.1.2 Adding Sites

You can add the sites via ShoreWare Director before configuring your dialing plan (or alternatively, you can configure your dialing plan and then add sites at a later time). To add a site via Director, follow the procedure below:

Step 1 Launch **ShoreWare 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 **Sites** link.

Step 4 Click on the **Add a new site in** drop-down menu and select the country where the site will be added.

Step 5 Click the **Go** link to display a window similar to the one shown below:

Figure 6-8 Add a new site

Step 6 Enter the name of the site, along with all other relevant information, in the appropriate fields. (Refer to the “ShoreTel Sites” chapter in the *ShoreTel Administration Guide* for additional information on con-figuring this window.)

The Extension Prefix field will not appear in this window until after you have modified the Dialing Plan window (which is our next task).

Step 7 Click **Save** to store your changes.

Step 8 Repeat this process to add any other sites that you would like to include in the dialing plan.

Once you have created the dialing plan and saved your dialing plan configurations, you can return to the Edit Sites window in Director to verify that the changes have been propagated throughout the system. By clicking on the name of the site, you will see an Extension Prefix field. The field should be populated with the value entered in the Extension Prefix Warning window, as shown in the window below:

Figure 6-9 Extension Prefix field now populated

6.5.1.3 Adding Users to the System

When the On-Net Dialing feature has been enabled and the extension prefix for a site has been updated, the first new user added to the system may not receive the site's new prefix. (This is due to cookies in the system populating the new user's extension with old and outdated information.) However, after this first user has been added, subsequent users will have their extensions automatically populated with the correct site prefix.

Details:

User numbers can vary in length from 3 to 5 digits. All user numbers in the system must be the same length.

6.6 Quick Reference of Star Codes

Certain features and functions can be performed via the telephone interface through the use of star codes. By pressing the star key (i.e. asterisk) on your phone’s keypad, followed by a combination of numbers, you can perform many tasks that would otherwise require the use of a soft key, option button, or programmable button.

6.6.1 Common Star Codes

Park a call	***11** + ext.
UnPark a call	***12** + ext.
Picking Up a Remote Extension	***13** + ext.
Picking Up the Night Bell	***14**
Using the Intercom	***15** + ext.
Barge In	***16** + ext.
Silent Monitor	***17** + ext.
Toggling the Hunt Group Status	***18** + Hunt Group ext.
Whisper Page	***19** + ext.
Changing Call Handling Mode and Forwarding	**VoiceMail** + password + **#** + **72**
Changing Extension Assignment	**VoiceMail** + password + **#** + **731**
Unassign Extension Assignment	**VoiceMail** + password + **#** + **732**
Assign Extension to External Number	**VoiceMail** + password + **#** + **733**

6.6.2 Extension Assignment Star Codes

Transfer a call	****** + destination + **# #**
Conference a call	****** + destination + ******
Hold a call	******
Hang up	**# #**
Access other "common" star codes	****** + *****star code (between 11 and 19) + ext.

6.6.3 Trunk Star Codes

Blocking and Caller ID — ***67** + ext.

- When a user places an external call, they can block their Caller ID using the "*67" command. The user dials the trunk access code, followed by *67, followed by the external number.
- When dialing in this manner, the call will be considered "non-routable" and will only access trunks at the local site. The number is dialed "as is" (i.e. as if a user dialed it). No digit manipulation will be performed.

Unblocking Caller ID — ***82** + ext.

- When a user places an external call, they can unblock their Caller ID delivering using the "*82" command. The user dials the trunk access code, followed by *82, followed by the external number.
- When dialing in this manner, the call will be considered "non-routable" and will only access trunks at the local site. The number is dialed "as is" (i.e. as if a user dialed it). No digit manipulation will be performed.

CHAPTER 7

Network Call Routing

This chapter provides an overview of call routing, and digit-manipulation capabilities of the ShoreTel system. The information in this chapter is particularly useful for administrators of larger, multisite installations.

7.1 Overview

When a phone number is dialed in a ShoreTel system, the system performs three distinct operations on telephone numbers:

Digit collection. Voice switches collect the digits in a telephone number.

Network call routing. After collecting the digits, the switch checks the number against a user's call permissions, adds trunks to the route list, and makes a final route decision for the call.

Digit manipulation. The switches manipulate the dialed numbers before outpulsing them to the service provider.

In this chapter you will learn how to plan your network call routing.

7.2 Checklist

Before configuring your phones (but **after** mapping out your network and trunk configuration), you need to review the topics in the table below:

Task	Description
Call Permissions	page 96
Account Codes	page 97
Trunk Availability	page 98
Specifying Parameters for the Routing Decision	page 99

Table 7-1 Network Call Routing Checklist

7.3 Define Network Call Routing

Once an external telephone number has been collected, the switching software checks the number against the user's call permissions, finds the list of available trunks, and then makes a routing decision based on several criteria.

7.3.1 Call Permissions

Each dialed number is compared against the user's call permissions. If the call is denied, the calling party will be routed to a "fast busy" intercept tone. If the call is allowed, the routing continues.

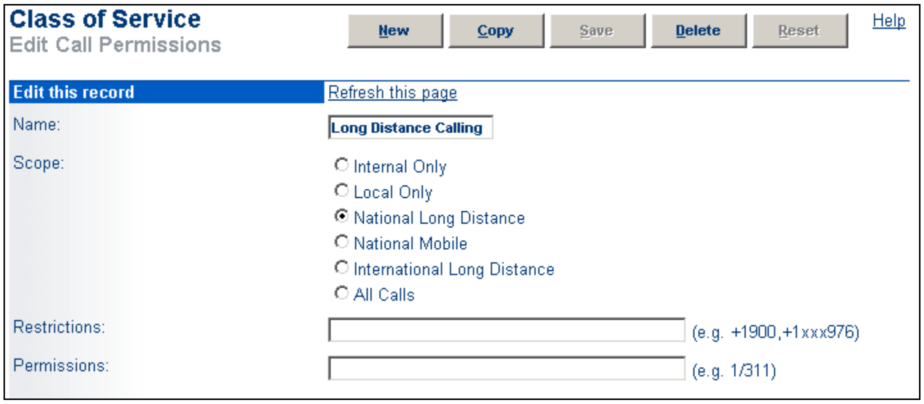

Figure 7-1 Call Permissions Edit Page

To define call permissions:

Step 1 Open the ***Call Permissions*** edit page (Figure 7-1).

Step 2 Select the ***Scope***. Scope allows you to set a general permission level and is presented from the most restrictive to the most permissive. The Restrictions and Permissions listed are applied in addition to the general scope setting for the Class of Service.

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. The call permission does not apply to any of the trunk group area codes.

National Long Distance also allows calls to long-distance numbers within the country, as defined on the ***Site*** edit page.

National Mobile allows calls to mobile phones in countries (e.g. Ireland) with "caller pays" billing plans.

International Long Distance also allows calls to international numbers, as defined on the ***Site*** edit page.

All Calls allows calls to any number, including 1900, Operator Assisted, and Carrier Select numbers, as well as use of Vertical Service Codes. This is the default.

Step 3 Enter restriction and permission rules. The Restrictions and Permissions listed are applied in addition to the general scope setting. The comma separated restriction expressions are limited to a total of 50 characters.

Follow these guidelines for entering restrictions:

In general, 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., use +1408. All 7-digit and 10-digit numbers must be entered in canonical format (+Country Code, Area Code, and Subscriber Number).

Non-routable calls (311, 411, etc.) for a country must be designated by the country code plus the "/" character. For example, to restrict 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.

To simplify the entering of call permissions, the wild-card character "x" can be used to represent any number. For instance, to block all calls to 976 prefixes, enter "+1xxx976" as a restriction.

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.

7.3.2 Account Codes

If Account Code Collection Service is enabled, when a user dials a number that is outside the scope of his or her call permissions, the call is automatically routed to the Account Code Collection Service extension on the HQ server. The Account Code Collection Service captures call details that can be reviewed in the call detail reports. For more information on these reports, see the ***ShoreTel Administration Guide***.

The collection of account codes is enabled on a per-user group basis and can be set to be one of three states: ***disabled***, ***optional***, or ***forced***.

The Account Code Collection Service is associated with a configurable extension and has a dedicated user group that defines ultimate call permissions and trunk group access.

When account code collection is enabled or forced for a member of the user group, calls placed via the telephone or the ShoreTel Communicator are first filtered by call permissions. Calls restricted by call permissions are automatically routed to the extension associated with the Account Code Collection Service. Upon receiving the call, the Account Code Collection Service prompts the user to enter an account code and press the pound (#) key.

If the user enters an account code that does not match the digits in a stored account code, the system plays a message explaining the problem and prompts the user to re-enter the account code. When the user enters an account code that matches one of the stored codes, the code is collected, and the call is completed.

Call Permissions specifies the dialed numbers that are directed to the Account Code Collection Service for any user groups configured for account codes.

Calls redirected to the account codes extension are completed using the trunk access and call permissions associated with the Account Code Collection Service.

The Account Code Collection Service examines outbound calls against two sets of permissions:

1. Checks call permissions for the caller's user group to determine if an account code must be collected.

2. If user group permissions specify the collection of an account code, a check is performed on the call permissions for the Account Code Collection Service to determine whether call will be permitted or rejected.

   If the call is rejected, the intercept tone is played.

The Account Code Collection Service is associated with a system extension hosted on a SoftSwitch that only runs on the headquarters (HQ) server.

If the Headquarters SoftSwitch is unavailable to the ShoreGear switch from which a call originates, the call is handled according to the permissions set for the caller's user group. Calls placed by users who are configured for ***optional*** account code collection are placed. Calls placed by users who are configured for ***forced*** account code collection are rejected.

Wildcard characters (represented with a question mark) can be used in place of DTMF digits in the account code. When wildcards are used, a length check is performed instead of a more thorough validation of the code. Although this reduces the stringency of the validation process, it allows the system to support far more than 50,000 account codes – the previous account code limitation.

Refer to the chapter on Call Control in the ***ShoreTel Administration Guide*** for more information about account codes and account code wildcards.

7.3.3 Trunk Availability

For a trunk to be included in the list of possible trunks that can be hunted, the following conditions must apply:

- The trunk must have an access code that matches the access code dialed.
- The trunk must be assigned to the user. (Trunk groups are assigned to user groups.)
- The trunk must be capable of the requested service (Local, Long Distance, International, n11, 911, Easily Recognizable Codes, Explicit Carrier Selection, and Operator Assisted). These services are defined on the ***Trunk Group*** edit page as shown in Figure 7-2.
- The trunk must be in service.
- The trunk must not already be in use.
- The trunk must be on a switch that the user's switch can reach. (The network is up and running.)
- For multisite calls, the admission control must be met at both sites. Admission control is defined on the ***Site*** edit page.
- If call is long distance from the trunk, it was not local to the caller. For example, network call routing will not send a local call via a trunk in another state.

To define trunk services:

Step 1 Open the ***Trunk Services*** dialog box on the ***Trunk Group*** edit page.

Step 2 Select the services that will be available for the selected trunk.

See the ***ShoreTel Administration Guide*** for more information about the ***Trunk Group*** edit page.

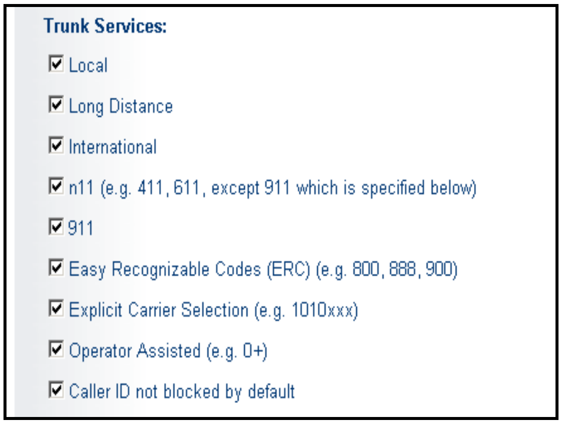

Figure 7-2 Trunk Services on the Trunk Group Edit Page

To define admission control:

Step 1 Open the ***Site*** edit page.

Step 2 Enter the proper amount in the ***Admission Control Bandwidth*** field.

See the ***ShoreTel Administration Guide*** for more information about the ***Site*** edit page and for instructions about computing Admission Control Bandwidth.

7.3.4 Specifying Parameters for the Routing Decision

Once the available set of trunks is established, the switching software makes a routing decision, with the goal of minimizing toll charges and WAN bandwidth. The Network Call Routing algorithm bases the routing decision on the Local Area Code, Additional Local Area Codes, and Nearby Area Codes defined on the ***Trunk Group*** edit page.

7.3.4.1 Network Call Routing Algorithm

When multiple trunks meet the same criteria, a trunk is seized randomly. In general, trunks that are configured last are hunted first. Over time, however, as trunks are deleted and added, hunting becomes increasingly random.

Digital trunks are given precedence over analog trunks in all routing decisions.

To make the routing decision, the algorithm poses the following questions. For the number dialed, is there:

1. A trunk at the originating site for which the call is local?
2. A trunk at the proxy site for which the call is local?
3. A trunk at any other site for which the call is local?
4. A trunk at the originating site for which the call is considered nearby?
5. A trunk at the proxy site for which the call is considered nearby?
6. A trunk at any other site for which the call is considered nearby?
7. A trunk at the originating site designated for long distance?
8. A trunk at any proxy site designated for long distance?
9. A trunk at any other site designated for long distance?

10. Any remaining trunk available at originating site?
11. Any remaining trunk available at the proxy site?

To specify parameters for the routing decision:

Step 1 Open the ***Network Call Routing*** page on the ***Trunk Group*** edit page, shown in Figure 7-3

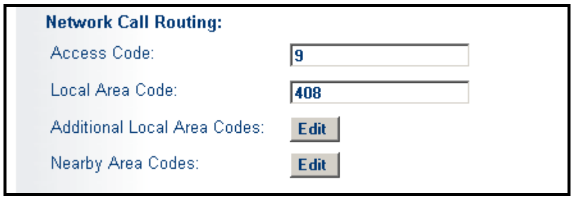

Figure 7-3 Network Call Routing on the Trunk Group Edit Page

Step 2 Enter values into the ***Local Area Code***, ***Additional Local Area Codes***, and ***Nearby Area Codes*** fields.

Step 3 Open the ***Trunk Group*** edit page and, toward the bottom of the page, click ***Go to Local prefixes***.

The ***Local Prefixes*** dialog box appears. It allows you to enter prefix exceptions against a local area code. The Network Call Routing algorithm handles prefix exceptions for the local area code as long distance, which minimizes toll charges.

See the ***ShoreTel Administration Guide*** for more information about the ***Trunk Group*** edit page and the ***Local Prefixes*** dialog box.

The area codes on the ***Site*** edit page have no impact on call routing decisions

CHAPTER 8

Telephone Planning and Ordering

This chapter provides information on the types of telephones supported by the ShoreTel system and what to consider when planning phones for your system.

8.1 Checklist

Review the following telephone planning topics before proceeding to the next chapter:

Task	Description
Application Considerations	page 101
Fax Machines and Modems	page 103
ShorePhone Telephones	page 104
Analog Phone Requirements	page 105

Table 8-1 Telephone Planning and Ordering Checklist

8.2 Recommendations

The following recommendations will assist you with planning, ordering, and installing your telephones:

Select your telephones based on user requirements, your wiring infrastructure, and system objectives.

Order your telephones early. If you need a large quantity of them, you will need to order them several weeks in advance.

Have your cabling contractor place and test all your telephones. Have the contractor unpack, assemble, place, and test every telephone so that you can avoid this simple but time-consuming task.

If the telephone you choose requires local power, make sure there is an available outlet at each location.

8.3 Application Considerations

8.3.1 General Users

Typically, most users will be satisfied with a standard desk telephone that has a speakerphone and mute button, and supports Caller ID and Message Waiting. ShoreTel IP phones are fully featured and appropriate for most uses. IP phones come with the ShoreTel features available on preprogrammed buttons, and they can be deployed in areas where there are no computers to run the ShoreTel Communicator -Personal Access.

8.3.2 Workgroup Agents and Supervisors

Because workgroup agents and supervisors typically spend large amounts of time on the telephone, they often like headsets. With the ShoreTel Communicator the user can control the telephone in Handsfree Mode and use the analog telephone and headset purely as a highly reliable method for carrying voice.

ShorePhone analog phones do not display Caller ID for calls forwarded from a workgroup or hunt group.

8.3.3 Operators

Operators typically answer and transfer large numbers of telephone calls throughout the day. Operators should be outfitted with a comfortable headset, and they should use the Handsfree Mode feature, which effectively turns off the dial tone. In this way, operators can use the ShoreTel Communicator to answer and transfer calls rapidly using their computer, without the need to touch the telephone.

If an operator is using one of the ShorePhone multiline models, the Automatic Off-Hook Preference feature allows the user to select which audio path (speakerphone or headset) is automatically activated when a call is placed or when an incoming phone call is received. The featured can be configured from Director, ShoreTel Communicator, or from the IP phone.

Operators may also benefit from the programmable buttons feature, which allows users to assign functions to the custom keys on the multiline phones, and on the BB24 device. The programmable buttons feature allows a user to assign the extension monitoring feature to one of the custom buttons. The Programmable Toolbars feature allows a system administrator to program common functions and operations to buttons in a user's ShoreTel Communicator window so that an operator can perform many common tasks (e.g. answer call, transfer call, invoke URL, etc.) at the click of a button.

Some operators will benefit from a cordless telephone or a cordless headset, which gives them greater mobility.

8.3.4 Receptionists

Receptionists are typically satisfied with a standard desk telephone that supports Caller ID and Message Waiting with a speakerphone and mute button.

8.3.5 Conference Rooms

Most conference rooms are best equipped with a speakerphone from a reputable manufacturer. Since conference rooms do not have a ShoreTel Communicator client, users may find the ShoreTel IP phone useful. The ShoreTel IP phone provides single-button access to features such as transferring and conferencing calls.

8.3.6 Lobby Phones

A cost-effective wall-mount, slim-line, or desk telephone is adequate for most lobby phones, hall phones, and the like. The IP110/115 models offer a cost-effective telephone that is ideal for use in lobbies, lounges, or other common areas.

8.3.7 Multi-line Phones

ShoreTel offers extension monitoring from an IP phone. With this feature, an administrative assistant or workgroup supervisor can monitor up to five system extensions. The extension monitor feature can be enabled for ShoreTel IP phones from the User edit pages of ShoreWare Director. For more information, see the ***ShoreTel Administration Guide***.

8.3.8 Teleworkers

Both analog and IP phones can be included in a ShoreTel system as remote phones. Analog phones require use of the Extension Assignment, while IP phones are supported by setting an IP address range through ShoreWare Director.

8.4 Fax Machines and Modems

The ShoreTel system supports fax machines and modems in the United States and Canada (and not elsewhere).

Fax and modem calls are more sensitive to network problems than voice conversations. The human ear does not notice a lost packet during a voice conversation, but when a packet is lost during a fax transmission the line may be dropped. During a modem call, a lost packet can cause a retransmission. In the worst case, fax machines and modems will not establish a connection or may drop the call altogether. In general, fax and modem calls work across a local area network, but work on wide area networks only when there is virtually no packet loss and little jitter.

The ShoreTel system automatically detects both fax and modem tones, and boosts the voice encoding to a higher value to increase throughput. (G.711 at 64 Kbps is recommended.) It also stops the nonlinear processing of the echo canceller and fixes the size of the jitter buffer to a preset level. In addition, for modems, the echo canceller is frozen or stopped, since the modems use their own network echo cancellers.

8.4.1 Fax Machines

Fax machines require a high-quality IP network for proper operation.

The ShoreTel system supports distinctive ringing for inbound calls: calls from external parties have the classic single ring, whereas calls from internal parties have a distinctive double ring. Some fax machines detect the ringing pattern before answering and might not answer internal calls because of the distinctive ring pattern. In particular, you must turn off the "Intelligent Ring Mode" on some Hewlett-Packard fax machines to receive calls from internal parties.

8.4.2 Modems

The ShoreTel system supports "moderate-use" modem applications on the system. This is generally considered to be modem calls up to 28.8 Kbps that do not last longer than 15 minutes. If your application demands greater performance, you should bypass the ShoreTel system or move your modem application to a pure IP-based solution.

The expected modem performance in different configurations is as follows:

> Analog connection speeds will not exceed 33.6 Kbps and could be lower. External factors, including poor-quality trunk lines, ISP limitations, and multiple analog-to-digital conversions in the network, can have a significant impact on connection speeds.

Modem calls demand a high-quality network with virtually no packet loss. Packet loss should not exceed 0.001%, which can be achieved on a local area network or in a wide area network using leased T1 facilities.

Analog trunk ports should not be used if a digital trunk (T1) is available, since performance will be limited to 28.8 Kbps maximum. Digital trunks should be used instead.

Connection speeds are significantly affected by multiple packet-to-circuit conversions (including modem calls from one ShoreTel system to another). If a T1 line is used, modems should be able to connect at K56Flex/V.90 or approximately 48 Kbps.

8.5 ShorePhone Telephones

Both analog and IP telephones are available from ShoreTel.

ShorePhone analog phones do not display Caller ID for calls forwarded from a workgroup or hunt group.

8.5.1 ShorePhone-AP100

The ShorePhone-AP100 telephone provides a cost-effective analog solution for business and includes a high-quality speaker telephone and a large display for caller information.

See Section 2.11.1 on page 35 for a complete description of the ShorePhone-AP100 telephone and the list of voice switches that support it.

8.5.2 ShorePhone-IP Phones

The ShorePhone IP phones are supported by ShoreGear voice switches. With ShoreTel IP phones, you create an end-to-end IP network, or a single-wire-to-the-desktop solution. The ShoreTel IP phone's intuitive user interface provides a high level of comfort when performing telephone operations.

The newer ShorePhone multiline models offer programmable buttons, making it easy for users to quickly and easily assign common operations to the buttons on their phones. Depending on the model of the IP phone, up to five extensions could be monitored with this feature.

Keep in mind that the "Copy Programmable Buttons" feature can be used to duplicate a programmable button configuration from one phone to another, saving you hours of tedious work as new users are added. Once the programmable buttons on one user's IP phone have been configured, the system administrator can leverage this existing configuration by copying the button profile to subsequent users' phones. (See "Copying Programmable Buttons Configurations" in the ***ShoreTel Administration Guide*** for more information.)

The ShorePhone multiline phones support the Automatic Off-Hook Preference feature, allowing users to select which audio path (speakerphone or headset) is automatically activated when a call is placed or when an incoming phone call is received.

Similarly, the multiline[1] models have improved support for the Plantronics CS50 wireless headset. Users can answer or end calls by pressing the activation button on their headset when they hear their phone ring. The 565g model offers support for use with some Bluetooth wireless headsets.

The ShorePhone-BB24 provides additional shortcut functions for users of the multiline phones, behaving like an additional set of 24 custom buttons. Additionally, it offers an Ethernet switch port, allowing connection of a PC to the back of the button box.

All ShorePhone IP models support the ability to load custom ring tones on the phone. The system administrator can load a pair of internal and external ring tones onto each user's phone. Each user can have a unique ring tone, and ring tones must be in the .wav file format. Please refer to the "Configuring IP Phones" chapter in the ***ShoreTel Administration Guide***.

See Section 2.11 on page 35 for a complete description of the ShorePhone IP telephones.

8.6 Analog Phone Requirements

The ShoreTel system supports standard analog 2500-type telephones, including the CLASS (Custom Local Area Signaling Services) features of Caller ID Name, Caller ID Number, and Message Waiting in the United States and Canada.

Outside the United States and Canada, the ShoreTel system supports the local standard analog telephones that support DTMF signaling. Analog Caller ID Number and Message Waiting are supported in the following countries:

- France
- Germany
- Italy
- Spain
- United Kingdom

Outside of the United States, Canada, and the countries mentioned in the bulleted list above, the features of Caller ID Name, Caller ID Number, and Message Waiting are not supported. See Appendix A, starting on page 263, for more information.

The following list summarizes key requirements for analog phones:

2500-type telephones: The ShoreTel system supports standard 2500-type telephones. (It does not support 500-type rotary telephones.)

DTMF signaling, even during power failure: The ShoreTel system uses DTMF tones for signaling with telephones and trunks. It is mandatory that the telephone support DTMF signaling even when power is interrupted, to allow users to make calls in emergency situations.

Flash button: A Flash button is required on analog phone sets to activate call control features from the telephone, including transfer, conference, pickup, and park. ShoreTel *does not* recommend using the hook switch to simulate the Flash button, since this can lead to accidental hang-ups.

If a speakerphone is required:

Mute button: Users in the enterprise typically demand that their speakerphone have a mute button. Since telephones are often designed with the residential market in mind, some speakerphones do not have a mute button, which may lead to end-user complaints.

1. IP560g and newer IP560 models support this feature while older IP560 models do not. To determine an IP560's compatibility, check the model number on the back of the phone. If the model number ends with a suffix of "-03" or higher, the phone supports this feature. If the suffix ends in "-01" or "-02" the feature is not supported by the phone.

If message waiting is required (United States and Canada only):

CLASS (FSK) message waiting indicator: CLASS message waiting–compatible telephones provide a highly reliable method for turning message waiting lights on and off.

Telephones that rely on a stutter dial tone to control the message waiting light are unreliable and should be avoided.

The ShoreTel system does not support telephones that use voltage-driven message waiting lights.

You should select telephones from a reputable manufacturer. Although most phones on the market are of good quality, ShoreTel recommends that you stay away from "clone" or "low-ball" manufacturers.

Here are some additional considerations:

Not too many buttons: Some telephones come with lots of complicated buttons and options that drive up the price of the device. The ShoreTel system delivers advanced features through desktop applications that are integrated with your enterprise tools. Telephones with lots of features and buttons are not necessary.

No answering machine: The ShoreTel system includes an integrated voice mail system for all users at all sites. Telephones with integrated answering machines are not necessary.

No hold button: Telephones with a hold button do not actually put the caller on system hold, so the caller will not hear music on hold or have the correct call control status details.

CHAPTER 9

Network Requirements and Preparation

Use the information in this chapter to determine specific network requirements for the ShoreTel system. After determining the network requirements, you will be ready to configure your network appropriately.

9.1 Checklist

Review the following planning topics before proceeding to the next chapter:

Task	Description
Advantages of Voice Over IP	page 108
Understanding the Requirements for Toll-Quality Voice	page 108
WAN Technology Choices	page 121
IP Address Assignment	page 122
Time Services	page 126
Virtual Private Network (VPN)	page 127
Firewalls	page 129
Media Encryption	page 131
Session Initiation Protocol (SIP)	page 132
Example Network Topologies	page 132
Computing Admission Control Bandwidth	page 133

Table 9-1 Network Requirements and Preparation Checklist

9.2 Overview

The ShoreTel system is an IP-based voice solution deployed across your IP network. This allows the components of the system to be located anywhere on your IP network, resulting in a single system for all your voice applications at all locations. This single system approach significantly reduces the complexity associated with legacy systems that consist of multiple PBXs, multiple voice mail systems, multiple auto-attendants, and multiple automatic call distribution systems, each with their unique management interfaces.

Since the ShoreTel system becomes another application on your IP network, it is important to understand how the system integrates with your data network. As you migrate your network to include voice as another application across your wide area network, it becomes necessary for your IP LAN and WAN to provide a network that meets the requirements for toll-quality voice. The ability of your network to deliver this performance will vary based on the number of simultaneous calls between locations, the voice quality required, and the other application traffic on the network. Some of the key considerations are:

- Bandwidth
- Service levels
- Addressing

9.3 Advantages of Voice Over IP

Going back to the basics of voice, consider a traditional call over the Public Switched Telephone Network (PSTN). The PSTN is a circuit-switched network. A telephone call reserves an end-to-end physical circuit for the duration of the call. This circuit consists of many subsegments within the PSTN, and a subsequent call between the same two endpoints may follow a different path. However, for the duration of the call, the circuit is fully available to that single call.

Packet-switched networks, such as the Internet, do not reserve a circuit between endpoints. Instead, messages or files are broken into many small packets. These packets may take different routes from source to destination, traveling along network circuits that are shared with packets from other sources. These packets travel to the final destination, where they are reconstructed into the original message or file.

One analogy between circuit-switched and packet-switched networks is that of railway versus roadway transportation systems. A railway is similar to a circuit-switched network. The path of the train is essentially reserved, and the whole train travels intact from source to destination. A roadway, on the other hand, is shared among many smaller units, each having the intelligence to find its destination. The railway provides a clear end-to-end path, relatively immune to delays, but at a high overhead cost. The roadway can be used more efficiently, but it is vulnerable to congestion.

The advantage of circuit-switched networks is that they provide dedicated bandwidth between endpoints and therefore can easily guarantee a known, consistent quality of service. Their disadvantage is their poor utilization of network resources, since they demand a dedicated, separate network relative to the packet-switched network. Conversely, the advantage of packet-switched networks is that they provide better utilization of network resources, enable flexible traffic routing, provide a single network to manage, allow for standard voice and data monitoring tools to be used, allow applications to be shared over a common network, and enable applications to become more portable—and this is just the beginning.

9.4 Understanding the Requirements for Toll-Quality Voice

The ShoreTel system has been designed to deliver the highest possible voice quality. In fact, third-party testing by Miercom has confirmed that the ShoreTel system provides both low latency and high voice quality.

With the superior design of the ShoreTel system, all that is needed to achieve toll-quality voice communications is to deploy the system over a properly designed network infrastructure. This section provides you with the background to understand the factors involved in engineering an IP network that is ready for voice communications.

In general, to ensure voice quality on the LAN, the ShoreTel system must be used in a switched Ethernet network. To ensure voice quality on the WAN, the ShoreTel system requires that you do the following:

Get a service level agreement (SLA) from your WAN service provider.

Using your routers, prioritize your voice traffic ahead of your data traffic.

Set the ShoreTel Admission Control feature to ensure that the voice traffic does not flood the WAN links.

With these items taken into consideration, you can simply and easily achieve toll-quality voice using the ShoreTel system.

The ShoreTel system has been designed to work in a multi-vendor network environment and therefore leverages standards to ensure voice prioritization.

IP Phone Supported Methods

Layer 2 IP Precedence (802.1p and 802.1q) (this only applies on the LAN)
Layer 3 Differentiated Services Code Point (DiffServ/ToS)
Layer 4 UDP 5004

ShoreGear Voice Switch Supported Methods

Layer 3 Differentiated Services Code Point (DiffServ/ToS)
Layer 4 UDP 5004

9.4.1 Network Requirements

When your voice traffic travels across your IP network, you must ensure that your network does all of the following:

Delivers enough bandwidth
Meets the latency and jitter requirements
Meets the packet loss requirements for toll-quality voice

You also need to prioritize your voice traffic over your data traffic and configure the ShoreTel system's Admission Control feature.

9.4.2 Bandwidth Requirements

The amount of bandwidth for voice calls depends on these details:

Number of simultaneous calls
Voice encoding scheme in use
Amount of signaling overhead

9.4.2.1 Voice Encoding

	Linear Broadband	Linear	G.711	ADPCM	G.729a
Sample rate	16 KHz	8 KHz	8 KHz	8 KHz	8 KHz
Effective sample size	16 bits	16 bits	8 bits	4 bits	1 bit
Data rate	256 Kbs	128 Kbps	64 Kbps	32 Kbps	8 Kbps
Supported end points	ShorePhones: IP110/115/212k/230/530/560/560g	All ShoreGear ShorePhones	All ShoreGear ShorePhones: IP110/115/212k/230/530/560/560g	All ShoreGear	All ShoreGear ShorePhones: IP110/115/212k/230/530/560/560g

Table 9-2 Voice Encoding

Within a site, linear broadband encoding is recommended, since bandwidth in the LAN is inexpensive and readily available. Between sites, G.729a is recommended because it uses the least amount of bandwidth. The linear codec provides slightly higher voice quality than G.711, but should not be used if there are any bandwidth concerns.

If you select linear broadband or linear encoding, end points that do not support either codec will negotiate for the highest quality codec for both end points, and G.711 is the only high-quality codec shared by all end points.

9.4.2.2 ShoreTel TCP and UDP Port Usage

ShoreTel uses the following TCP and UDP ports for traffic.

Originating Device	Traffic Type	Destination Device						
		Switch	IP Phone	Desktop	SoftPhone	DVM Server	HQ/Director Server	Other
Switch	Call Control	UDP 5440 Location Service UDP 5441 Call Control UDP 5443 Bandwidth Manager UDP 5445 Admission Control	UDP 2427 MGCP		UDP 2427 MGCP	TCP 1024-65535 RPC - NCC UDP 2727 MGCP - Media proxy UDP 5440 Location Service UDP 5441 Call Control UDP 5442 DRS UDP 5443 Bandwidth Manager UDP 5445 Admission Control UDP 5446 DRS keepalive	TCP 1024-65535 RPC - NCC UDP 2727 MGCP - Media proxy UDP 5440 Location Service UDP 5441 Call Control UDP 5442 DRS UDP 5443 Bandwidth Manager UDP 5445 Admission Control UDP 5446 DRS keepalive	UDP 5060 SIP
Switch	Media Stream	UDP 5004 RTP (dynamic: 1024-65535)	UDP 5004 RTP (dynamic: 1024-65535)		UDP 5004 RTP (dynamic: 1024-65535)	UDP 5004 RTP (dynamic: 1024-65535)	UDP 5004 RTP (dynamic: 1024-65535)	UDP 1024-65535 RTP - for SIP
Switch	RPC Connection Negotiation					TCP 111 RPC Port Mapper UDP 111 RPC Port Mapper	TCP 111 RPC Port Mapper UDP 111 RPC Port Mapper	

Table 9-3 Port Usage

Originating Device	Traffic Type	Destination Device						
		Switch	**IP Phone**	**Desktop**	**SoftPhone**	**DVM Server**	**HQ/Director Server**	**Other**
Switch	Configuration Control					TCP 21 FTP CTL - Boot files TCP 20 FTP DATA - Boot files	TCP 21 FTP CTL - Switch Boot TCP 20 FTP DATA - Switch Boot	UDP 67 DHCP Server
Switch	Maintenance							UDP 162 SNMP TRAP
IP Phone	Call Control	UDP 2727 MGCP	UDP 5554 BB to Phone					
IP Phone	Media Stream	UDP 5004 RTP [if dynamic 1024-65535]	UDP 5004 RTP [if dynamic 1024-65535]		UDP 5004 RTP [if dynamic 1024-65535]	UDP 5004 RTP if dynamic 1024-65535]		
IP Phone	Configuration Control					TCP 21 FTP CTL - Config TCP 20 FTP DATA - Config ICMP PING	TCP 2 FTP CTL - Config TCP 20 FTP DATA - Config ICMP - PING UDP 5004 RTP [if dynamic 1024-65535]	UDP 67 DHCP Server UDP 123 SNTP
Desktop	Call Control					TCP 1024-65535 MS RPC - Remote TAPI	TCP 1024-65535 MS RPC - Remote TAPI	
Desktop	Configuration Control					TCP 5440 CSIS	TCP 80 HTTP Web client, Online help TCP 5440 CSIS	
Desktop	RPC Connection Negotiation					TCP 135 MS RPC Port Mapper	TCP 135 MS RPC Port Mapper	
SoftPhone	Call Control	UDP 2727 MGCP						
SoftPhone	Media Stream	UDP 5004 RTP [if dynamic 1024-65535]	UDP 5004 RTP [if dynamic 1024-65535]		UDP 5004 RTP [if dynamic 1024-65535]	UDP 5004 RTP [if dynamic 1024-65535]	UDP 5004 RTP [if dynamic 1024-65535]	
SoftPhone	Configuration Control						TCP 80 HTTP	
DVM Server	Call Control	TCP 1024-65535 RPC - NCC UDP 2427 MGCP - Media proxy UDP 5440 Location Service UDP 5441 Call Control UDP 5443 Bandwidth Manager UDP 5445 Admission Control		TCP 1024 65535 Remote TAPI		TCP 1024-65535 MS RPC - DTAS/TMS TCP 5441 Call data UDP 5440 Location Service UDP 5441 Call Control UDP 5443 Bandwidth Manager UDP 5445 Admission Control UDP 5446 DRS keepalive	TCP 1024-65535 MS RPC - DTAS/TMS TCP 1024-65535 MS RPC - DB access TCP 5441 Call data UDP 5440 Location Service UDP 5441 Call Control UDP 5443 Bandwidth Manager UDP 5445 Admission Control UDP 5446 DRS keepalive	

Table 9-3 Port Usage

Originating Device	Traffic Type	Destination Device						
		Switch	**IP Phone**	**Desktop**	**SoftPhone**	**DVM Server**	**HQ/Director Server**	**Other**
DVM Server	Media Stream	UDP 5004 RTP [if dynamic 1024-65535]	UDP 5004 RTP [if dynamic 1024-65535]		UDP 5004 RTP [if dynamic 1024-65535]			
DVM Server	RPC Connection Negotiation	TCP 111 RPC Port Mapper UDP 111 RPC Port Mapper				TCP 135 MS RPC Port Mapper	TCP 111 RPC Port Mapper UDP 111 RPC Port Mapper TCP 135 MS RPC Port Mapper	
DVM Server	Configuration Control	TCP 1024-65535 Firmware download						
DVM Server	Maintenance							TCP 1024-65535 RPC - Quicklook
DVM Server	Distributed Voice Mail					TCP 25 SMTP - Voice Mail transport	TCP 25 SMTP - Voice Mail transport	
DVM Server	Voice Mail Notification							TCP 25 SMTP
HQ/ Director Server	Call Control	TCP 1024-65535 RPC - NCC UDP 2427 MGCP - Media proxy UDP 5440 Location Service UDP 5441 Call Control UDP 5443 Bandwidth Manager UDP 5445 Admission Control		TCP 1024-65535 Remote TAPI		Call Control TCP 1024-65535 MS RPC - DTAS/ TMS TCP 1024-65535 MS RPC - DB Notify TCP 5441 Call data UDP 5440 Location Service UDP 5441 Call Control UDP 5443 Bandwidth Manager UDP 5445 Admission Control UDP 5446 DRS keepalive		
HQ/ Director Server	Media Stream	UDP 5004 RTP [if dynamic 1024-65535]	UDP 5004 RTP [if dynamic 1024-65535]		UDP 5004 RTP [if dynamic 1024-65535]			
HQ/ Director Server	RPC Connection Negotiation	TCP 111 RPC Port Mapper UDP 111 RPC Port Mapper				TCP 111 RPC Port Mapper UDP 111 RPC Port Mapper TCP 135 MS RPC Port Mapper		
HQ/ Director Server	Configuration Control	TCP 1024-65535 Firmware download						
HQ/ Director Server	Maintenance	TCP 5555 Diagnostic ipbxctl	TCP 5555 Diagnostic phonectl			TCP 1024-65535 RPC - Quicklook		

Table 9-3 Port Usage

Originating Device	Traffic Type	Destination Device						
		Switch	IP Phone	Desktop	SoftPhone	DVM Server	HQ/Director Server	Other
HQ/ Director Server	Distributed Voice Mail					TCP 25 SMTP - Voice Mail transport		
HQ/ Director Server	Voice Mail Notification							TCP 25 SMTP
HQ/ Director Server	CDR							TCP 3306 - CDR archive on remote server
Other	Call Control	UDP 5060 SIP						
Other	Media Stream	UDP 1024-65535 RTP - for SIP						
Other	Configuration Control	UDP 68 DHCP Client	UDP 68 DHCP Client				TCP 80 HTTP - Director	
Other	Maintenance	TCP 23 Telnet UDP 161 SNMP	TCP 23 Telnet			TCP 80 HTTP - Quicklook		

Table 9-3 Port Usage

9.4.2.3 Bandwidth in the LAN

For LAN calls using the voice switches, 10 msecs of voice samples are encapsulated in a Real Time Protocol (RTP) packet before being transmitted onto the LAN. For IP phones and SoftPhones, 20 msecs of voice samples are encapsulated in an RTP packet before being transmitted onto the network.

The protocol overhead consists of 12 bytes for the RTP header, 8 bytes for the UDP header, 20 bytes for the IP header, and 26 bytes for the Ethernet framing. When ADPCM voice encoding is used, an additional 4 bytes are added to the voice data for decoding purposes. This yields an effective LAN bandwidth as shown in Table 9-4.

	Linear Broadband	Linear	G.711	ADPCM	G.729a	G.729a
Voice data (10 msec)	320	160	80	40+4[a]	20 (20 msec)[b]	30 (30 msec)
RTP header	12	12	12	12	12	12
UDP header	8	8	8	8	8	8
IP header	20	20	20	20	20	20
Ethernet header and framing[c]	26	26	26	26	26	26
Total bytes per packet[d]	386	226	146	110	86 (20 msec)	96 (30 msec)
Bandwidth for voice only[e]	256 Kbps	128 Kbps	64 Kbps	32 Kbps	8 Kbps	8 Kbps
Bandwidth with overhead	309 Kbps	181 Kbps	117 Kbps	88 Kbps	34 Kbps	34 Kbps

Table 9-4 LAN Bandwidth—Bytes

a. When ADPCM using voice encoding, four bytes are added to the voice data for decoding purposes.
b. G.729a is supported in 10-msec, 20-msec, and 30-msec packets.
c. Ethernet framing = 14 bytes of Ethernet header, a 4-byte checksum, and 8 bytes of additional framing.
d. Voice data bytes per packet = (# bits/sample) x (8 samples/msec) x (10 msecs/packet) / (8 bits/byte).
e. Bandwidth = (# bytes/10 msecs) x (8 bits/byte).

For calls between analog telephones, voice bandwidth is used only on the connection between the voice switches. For calls involving IP telephones, the bandwidth is required between the IP phones at the user's desktop. This means that for IP telephones, network planning must include provisioning capacity to each IP phone.

When SIP is not enabled, RTP traffic is always sent to UDP port 5004. The source port is random. When SIP is enabled, both the source and destination ports are random.

9.4.2.4 Bandwidth in the WAN

Increasing the number of voice samples per packet decreases the bandwidth required (since the percentage of signaling overhead is reduced); however, it also increases the latency of the voice call, which can result in poorer voice quality. Consequently, the ShoreTel system uses 10-msec voice packets on the LAN, where bandwidth is readily available, and 20-msec voice packets on the WAN, where bandwidth conservation is more important. WAN calls are calls made between ShoreTel system sites.

For WAN calls, routers with RTP Header Compression (cRTP) reduce the 40 bytes in the IP + UDP + RTP header to 4 bytes. If you want to use cRTP, make sure the router's implementation of cRTP does not increase the latency or jitter of the voice traffic, since this can have a negative impact on voice quality. If the router does increase latency or jitter with cRTP, add this to your overall expected latency and make sure you still have sufficient performance for acceptable voice quality.

Table 9-5 shows the resulting effective WAN bandwidth. It does not include the overhead associated with the underlying WAN network protocol, such as HDLC, frame relay, ATM, and VPN; however, the ShoreTel admission control software computes bandwidth requirements according to the data in this table and assumes a PPP header-size for computations.

	Linear Broadband	Linear	G.711	ADPCM	G.729a	G.729a
Voice data (20 msec)	640	320	160	80+4[a]	20	30
RTP header	12	12	12	12	12	12
UDP header	8	8	8	8	8	8
IP header	20	20	20	20	20	20
PPP header	5	5	5	5	5	5
Total bytes per packet[b]	685	365	205	129	65	75

Table 9-5 WAN Bandwidth—Bytes

	Linear Broadband	Linear	G.711	ADPCM	G.729a	G.729a
Bandwidth for voice only[c]	256 Kbps	128 Kbps	64 Kbps	32 Kbps	8 Kbps	8 Kbps
Bandwidth including overhead	284 Kbps	146 Kbps	82 Kbps	52 Kbps	26 Kbps	26 Kbps
Bandwidth after cRTP	260 Kbps	132 Kbps	68 Kbps	37 Kbps	12 Kbps	12 Kbps

Table 9-5 WAN Bandwidth—Bytes

a. When ADPCM voice encoding is used, an additional 4 bytes are added to the voice data for decoding purposes.
b. Voice data bytes per packet = (# bits/sample) x (8 samples/msec) x (20 msecs/packet) / (8 bits/byte)
c. Bandwidth = (# bytes/20 msecs) x (8 bits/byte)

9.4.3 Latency

Latency is the amount of time it takes for one person's voice to be sampled, digitized (or encoded), packetized, sent over the IP network, de-packetized, and replayed to another person. This one-way delay, from "mouth-to-ear," must not exceed 100 msecs for toll-quality voice, or 150 msecs for acceptable-quality voice. If the latency is too high, it interferes with the natural flow of the conversation, causing the two parties to confuse the latency for pauses in speech. The resulting conversation is reminiscent of international calls over satellite facilities.

The latency introduced by the ShoreTel system can be understood as follows: When a person talks, the voice is sampled by the ShoreGear voice switch, generating a latency of 5 msecs. If the call does not traverse ShoreTel voice switches and is handled completely internally by the switch, the latency is generated by the basic internal pipeline of the switch. In this case, the switch samples the voice, processes it, combines it with other voice streams (switchboard), and then converts it back to audio for output to the phone in 5-msec packets, for a total latency of about 17 msecs.

When the call transfers between voice switches, the voice is packetized in larger packets—10-msec for LAN and 20-msec for WAN—to reduce network overhead. The larger packets take more time to accumulate and convert to RTP before being sent out. On the receive side, the incoming packets are decoded and placed in the queue for the switchboard. For a 10-msec packet, this additional send/receive time is approximately 15 msecs, and for a 20-msec packet it is about 25 msecs.

For IP phones, the latency is 20 ms in the LAN and 30ms in the WAN.

When the codec is G.729a, the encoding process takes an additional 10 msecs and the decoding process can take an additional 10 msecs.

See Table 9-6 for specific information about latency on the ShoreTel system.

Configuration	Overhead	Encoding	Frame Size -5	Jitter Buffer[a]	Decoding	Total (+/– 5 msec)[b]
Switch	17	0	0	Varies	0	17

Table 9-6 Latency

Configuration	Overhead	Encoding	Frame Size -5	Jitter Buffer[a]	Decoding	Total (+/– 5 msec)[b]
LAN	17	5	5	Varies	5	32 + Jitter Buffer
WAN	17	5	15	Varies	5	42 + Jitter Buffer
G.729a (LAN and WAN)	17	15	15	Varies	15	62 + Jitter Buffer

Table 9-6 Latency

a. The jitter buffer varies, depending on network conditions. See below for more information.
b. If a call comes in on a trunk through either T1/E1 or analog loop-start, the total latency is increased by the delay in the PSTN. You must add this latency to the total latency. Latency for the PSTN varies; however, it is probably a minimum of 10 msecs (for local), and it could be as high as hundreds of msecs (for long international calls).

9.4.4 Jitter for Voice Switches

Jitter is the variation of latency across the network and the variation in packet processing inside the switches. To compensate for jitter, the ShoreGear voice switches continuously measure the jitter in the system and dynamically change the size of the receive jitter buffers to optimize voice quality.

If the jitter buffer is too small, there can be packet loss from buffer underflows. This occurs when the jitter buffer runs out of valid voice samples. If the jitter buffer is too large, there will be unnecessary latency. Both conditions have a negative impact on voice quality.

The jitter buffer starts at the minimum size of 0 msecs as packets from the network are placed into the switchboard queue for immediate processing. When jitter is detected on the network, the jitter buffer dynamically increases in increments of 5 msecs to compensate for increased jitter and decreases in size in reaction to less jitter. The maximum value of the jitter buffer is set by ShoreWare Director and ranges from 20 to 300 msecs, with a default of 50 msecs.

As the jitter increases on the network and the jitter buffer needs to be increased to guarantee timely audio play, the latency of the audio also increases. The system attempts both to maintain a minimum jitter buffer size that provides good-quality voice without dropping packets and to provide minimum latency.

For IP phones that are configured into the ShoreTel system, the jitter buffer is not configurable. The minimum jitter buffer is 10 msecs, and the maximum is 80 msecs.

Maximum values greater than 100 should rarely be necessary. If needed, this could indicate a problem in your network that should be addressed in another way.

9.4.5 Packet Loss

Lost packets can occur on the IP network for any number of reasons. Packet loss above 1% begins to adversely affect voice quality. To help reduce this problem, the ShoreGear voice switches have a feature called lost packet concealment. When there is no voice sample to be played, the last sample available is replayed to the receiving party at a reduced level. This is repeated until a nominal level is reached, effectively reducing the clicking and popping associated with low levels of packet loss.

Fax and modem calls demand essentially zero packet loss to avoid missing lines on fax calls and to avoid dropped modem calls. In addition, fax and modem calls, when detected, may change to a higher-rate codec.

9.4.5.1 Summary of the Network Requirements

Table 9-7 summarizes the network requirements for bandwidth, latency, jitter, and packet loss.

Parameter	Requirement
Bandwidth (WAN)	With ADPCM and no RTP Header Compression: **52 Kbps** per call With G.729a and no RTP Header Compression: **26 Kbps** per call With G.711 and no RTP Header Compression: **82 Kbps** per call **If your network uses VPN, bandwidth use is affected.**
Latency and jitter for toll quality	< 100 msecs total 100 msecs less 42 msecs allocated for the ShoreTel system yields a **58 msec budget** for the network. When G.729a encoding is used, 100 msecs less 62 msecs allocation for the ShoreTel system yields a 38 msec budget for the network.
Latency and jitter for acceptable quality	< 150 msecs total 150 msecs less 42 msecs allocated for the ShoreTel system yields a **108 msec budget** for the network. When G.729a encoding is used, 150 msecs less 62 msecs allocated for the ShoreTel system yields an 88 msec budget for the network.
Packet loss	< 1% for voice calls, and no packet loss for fax and modem calls

Table 9-7 Network Requirements

9.4.5.2 Impact of Long Network Outages

The ShoreTel system is a completely distributed system in which each ShoreGear voice switch provides all call control functionality for inbound and outbound calls, as well as features such as transfer, conference, pickup, and trunk selection. When there is a long network outage, the switches will detect the problem and run isolated from the switches that can no longer be reached. In the ShoreTel system, switches communicate every 30 seconds and disconnect when there is no response after 60 seconds.

9.4.6 Bandwidth Management

In addition to the network requirements discussed above, bandwidth management techniques need to be deployed to ensure that real-time voice data is not affected by bursts or high amounts of data traffic.

9.4.6.1 Local Area Network

To manage bandwidth in the local area network (intra-site) and meet the requirements for toll-quality voice, use Ethernet switching. Ethernet switching is cost effective and simple to provision. Your LAN configuration requirements will vary depending on your infrastructure and whether your network includes IP phones.

IP phones sample the user’s voice and convert the voice signal to IP packets using the Real Time Protocol (RTP). These packets must be tagged for higher prioritization in the network. ShoreTel IP phones have embedded Ethernet switches and will automatically prioritize voice traffic ahead of any data traffic coming from daisy-chained personal computers (for example, large files transfers and email).

On the local area network, there are several methods to prioritize voice packets, including:

IP Precedence = 5 (configurable, recommendation is 5)

DiffServ/ToS = EF (configurable, recommendation is EF)
UDP (Dest. port) = 5004 (when not using SIP)

The Ethernet switch infrastructure should be configured to prioritize traffic using one of these methods. This allows the voice traffic arriving at the switch to travel ahead of the data traffic.

ShoreTel customers typically choose to prioritize DSCP since this configuration is easy to set up on smart Ethernet switches.

When IP phones are used, the desktop connection to the user's computer and phone must also be part of your switched Ethernet network. The user's phone is connected to the port on the Ethernet switch, and the user's computer or other data device is connected to the integrated two-port Ethernet switch inside the IP phone. In this configuration, the switch port connected to the phone must be configured to prioritize the voice packets from the phone above the data packets.

PCs connected through IP phones will lose their connection to the network if the IP phone loses power.

Voice quality can be guaranteed by putting each of the ShoreGear voice switches and the ShoreWare server on its own Ethernet switch port. A network with this topology meets the bandwidth, jitter, and latency requirements for toll-quality voice without the additional need for special prioritization of voice packets.

9.4.6.2 Virtual LANs

An alternative method to prioritize voice over data is to create a separate virtual LAN strictly for your voice traffic. The ShoreTel IP phone as well as the ShoreGear voice switches can be configured on a specific VLAN.

Set the voice VLAN for higher prioritization in the network. The Ethernet switch infrastructure needs to be configured to prioritize the voice VLAN. This allows the voice traffic arriving at the switch to travel ahead of the data traffic.

9.4.6.3 Wide Area Network

To manage bandwidth in the wide area network, prioritize your voice traffic ahead of your data traffic. The voice packets on the ShoreTel system always travel on UDP port 5004, so you simply prioritize this UDP port within your routers with priority queueing. You can prioritize based on the voice switch IP address, the MAC address, or the physical port on the Ethernet switch. As an additional step, you can also prioritize the distributed call control signaling that always travels on UDP port 5440 through UDP port 5445.

If the voice traffic for the call needs to flow across a WAN link, the routers needs to be configured to prioritize voice ahead of data using one of the two tagging methods, DiffServ/ToS or UDP 5004.

ShoreTel customers typically choose to prioritize UDP 5004 to avoid costly network upgrades since older routers and more Ethernet switches support this function. Additionally, configuring UDP 5004 for prioritization is easy to set up.

9.4.6.4 Client Bandwidth

ShoreTel Communicators communicate with the ShoreWare server for call information and control, configuration changes, and advanced services such as extension monitoring. Table 9-8 provides an estimate of the client bandwidth used for each of the ShoreTel Communicator applications.

ShoreTel Communicator	Bandwidth Use
Personal	.2 Kbps
Professional	.2 Kbps
Operator	.2 Kbps + 1.5 Kbps
Extension Monitor	1.5 Kbps per monitored extension
Workgroup Agent	.25 Kbps
Queue Monitor	6.5 Kbps per queued call
Workgroup Supervisor	.25 Kbps
Queue Monitor	6.5 Kbps per queued call
Agent Monitor	1.5 Kbps per agent

Table 9-8 Typical ShoreTel Communicator Bandwidth Use

9.4.7 Distributed Call Control Signaling

Voice switches maintain communication with each other. A single voice switch maintaining basic connectivity with 59 other voice switches consumes less than 1.5 Kbps of bandwidth.

9.4.8 Admission Control in the Wide Area Network

To ensure that your voice traffic does not overwhelm the wide area network and degrade voice quality, the ShoreTel system has an Admission Control feature. From ShoreWare Director, you can limit the amount of WAN bandwidth used for telephone calls on a per-site basis. For a telephone call to be established between sites, admission control must be met at both sites. If the admission control limit is reached at a site, additional calls cannot be placed to or from the site, thus ensuring the voice quality of calls already in progress. If the user is making an outbound call, the call is automatically routed out of a trunk at the site. When making an extension-to-extension call, the user is informed that there is insufficient network bandwidth to complete the call. The user can try again later or dial the external number of the other user.

If PSTN failover is enabled for a user extension, the user's extension-to-extension calls are automatically routed to the public switched telephone network (PSTN) when there is insufficient bandwidth for an IP connection to phone.

9.4.9 Spanning Tree Protocol

Spanning Tree Protocol (STP) is used by Ethernet switches and routers to determine if there are multiple paths on the network between any two endpoints. You must disable STP on any network port that has a ShoreGear switch or ShoreWare server connected.

9.4.10 Traffic Shaping to Reduce Bottlenecks

Given that more applications are requiring WAN bandwidth, the need to optimize is increasingly important. This is particularly true for enterprises that want to deploy voice over virtual networks where quality of service and traffic shaping are required. With traffic shaping, it is possible to set policies that determine who or what gets top priority. For example, by prioritizing the various flows of traffic, an administrator can make sure that UDP (voice) traffic gets a higher priority than FTP (file download) traffic.

9.4.11 Echo Cancellation

Echo in a voice communication system is caused by signal reflections generated by the electrical circuits called hybrids that convert between two-wire (shared transmit and receive pair) and four-wire circuits (separate transmit and receive pairs). These reflections cause the speaker's voice to be heard in the speaker's ear as delayed by many milliseconds. Echo is present even in the traditional circuit-switched telephone network, but since the delay in a local circuit-switched call is so low, the echo is not perceivable. On a packet-based voice network, there is more delay, and the speaker may perceive the echo if it is not properly cancelled.

The DSP software on the ShoreGear voice switches provides dynamic echo cancellation. When a user places an extension-to-trunk call using an analog trunk on a ShoreGear voice switch, the user's voice bounces off the initial four-wire to two-wire conversion in the analog trunk circuit, then off the two-wire to four-wire in the central office, and finally off the called party's telephone. This echo returns from the central office and is cancelled by the echo canceller on the trunk port of the voice switch. The echo from the called party's phone, however, is usually cancelled or suppressed by the central office. If this echo is not cancelled, the user may hear himself or herself talking.

In the opposite direction, the external person's voice bounces off the user's telephone. This echo returns from the telephone and is cancelled by the echo canceller on the telephone port of the voice switch. If this echo is not cancelled, the external party hears himself or herself talking. This same process of echo cancellation applies to extension-to-extension as well as trunk-to-trunk calls.

ShoreGear switches can cancel echo received up to 16 msecs after being sent.

9.4.12 Resultant Voice Quality

As stated earlier, the ShoreTel system has been recognized for excellent voice quality. This is a result of the excellent hardware and software design that minimizes latency and dynamically adapts to the effects of jitter, packet loss, and echo introduced by the network.

There are two subjective testing methods that are used to evaluate voice quality. A method called Mean Opinion Score (MOS) is an open test in which a variety of listeners judge the quality of a voice sample on a scale of 1 (low) to 5 (high). There is general industry agreement on the theoretical maximum MOS value on a per codec basis that can be achieved (see Table 9-9).

Codec	Data Rate (Kbps)	MOS
Linear	128	4.5
G.711	64	4.1
ADPCM	32	3.85
G.729a	8	3.85

Table 9-9 Theoretical MOS Maximum Scores

Both the MOS test method and an interactive test method were used by Miercom. The interactive test focused on the conversational quality of the call. The results are shown in Table 9-10. The ShoreTel MOS scores are higher than the industry-standard values. This is likely a result of the subjective nature of the head-to-head test, which scores a relative ranking rather than an absolute ranking.

Codec	Data Rate (Kbps)	MOS	Interactive
Linear	128	Not tested	Not tested
G.711	64	4.46–4.87	4.66
ADPCM	32	3.96–4.05	4.33
G.729a	8	Not tested	Not tested

Table 9-10 ShoreTel MOS and Interactive Test Results

9.5 WAN Technology Choices

9.5.1 Minimum Bandwidth Requirements

The minimum WAN bandwidth required to deploy a voice switch at a site depends on the number of calls expected. With ADPCM, a single call consumes 52 Kbps, and if this call becomes a conference call, another 52 Kbps is needed, yielding a total of 104 Kbps. From a broadband perspective, the first available technology is 128 Kbps (ISDN), which leaves only 24 Kbps for other IP traffic. For teleworking applications, where only a single call is needed, 128 Kbps can be used. For other sites on the voice network, the minimum bandwidth recommended is 384 Kbps.

Various technologies are available from different service providers to provide IP connectivity between locations, as shown in Table 9-11.

Technology	Upstream Bandwidth Kbps	Downstream Bandwidth Kbps	Calls with ADPCM[a]
T1	1544	1544	26
Frame Relay	Varies	Varies	Varies
SDSL	1544	1544	26
SDSL	1024	1024	17
SDSL	768	768	13
SDSL	512	512	8
SDSL	384	384	6
IDSL	144	144	1 call only
ADSL	128	1,000 (varies)	1 call only
Cable	128 (varies)	1,000 (varies)	1 call only
ISDN BRI	128	128	Not supported
Dial-up modem	28.8–56	28.8–56	Not supported

Table 9-11 IP Connectivity Chart

a. Your bandwidth will vary, based on the WAN overhead for your particular system.

9.5.2 Leased T1

Leased T1 facilities are the most robust WAN technology available. Leased T1s are point-to-point links that inherently meet the network requirements for toll-quality voice since no ISP is involved. Dedicated T1s are priced on a per unit distance basis, making this a very cost-effective option over short distances.

9.5.3 Frame Relay

Frame Relay is a viable option as long as you get a committed information rate (CIR) that meets the bandwidth and network requirements for toll-quality voice communications.

9.5.4 SDSL

SDSL is considered "business-to-business" DSL in which you can negotiate a service level agreement with the service provider. Unlike T1, SDSL is priced on a flat bandwidth basis, making the price "distance insensitive" and cost-effective over long distances.

Although this is an excellent option, especially moving forward, ShoreTel has found the use of SDSL challenging, since the service providers often commit to a Service Level Agreement (SLA) they cannot fulfill. Many service providers have grown very fast, and the IP network is a patchwork of devices. These service providers are usually geared toward providing bandwidth for typical data applications, and a voice application highlights weaknesses in their network. Only with joint troubleshooting of the service provider's network, using tools such as ping plotters, has ShoreTel been able to achieve the SLA the service provider promised.

9.5.5 IDSL

IDSL modems, which have an uplink and downlink speed of 144 Kbps, can be considered for teleworking applications. The actual performance will vary based on your service provider and your applications.

9.5.6 ADSL

ADSL modems, which have an uplink speed of 128 Kbps, can be considered for teleworking applications. The actual performance will vary based on your service provider and your applications.

9.5.7 Cable Modems

Cable modems, which can have an uplink speed of 128 Kbps, can be considered for teleworking applications. The actual performance will vary based on your service provider and your applications.

9.5.8 ISDN BRI

ISDN BRI is not supported at this time.

9.5.9 Dial-Up Modems

Because of their inherent latency and low bandwidth, dial-up modems are not supported.

9.6 IP Address Assignment

Each ShoreGear voice switch requires one IP address. Each software server must be configured with a static IP address. You can use one of the following to serve an IP address to a voice switch:

- DHCP on a network server

The BOOTP server integrated into ShoreWare Director

The maintenance port on the front of the ShoreGear switches that provide a maintenance port. Refer to Appendix G, starting on page 303, for the location of the Maintenance port on ShoreGear switches.

If a voice switch has been configured to request a dynamic IP address, it puts a DHCP/ BOOTP request on the network when powered on. If the voice switch receives a response, it uses the new IP address. If no response is received, it reverts to the previous IP address. If there is no previous IP address, the voice switch continues trying to get an IP address.

If you use a DHCP server on the network, ShoreTel recommends that you configure reserved IP addresses such that the IP addresses of the voice switches do not inadvertently change.

If you do not have a DHCP server on the network, you can use the BOOTP server integrated into ShoreWare Director to assign IP addresses. ShoreTel does not support running DHCP on the ShoreWare server for serving either ShoreGear voice switches or other equipment.

The maintenance port can be used to configure the networking parameters.

The following recommendations will assist you with IP address assignment:

Ensure there is only one DHCP server on the network. If you have multiple DHCP servers on the network, you risk giving the voice switches an errant IP address that will remove the voice switches from service until the problem is corrected.

The ShoreTel system must be on a private network in some situations and on a public network in other instances. For example, if the enterprise is using a firewall with Network Address Translation (NAT), all remote facilities must establish VPN connections to the headquarters and be on the same private network. If the enterprise is not using NAT but is using firewalls, all remote locations must use public IP addresses.

Each IP telephone must be configured with a single unique IP address. You can configure the IP telephone through DHCP or manually on the telephone.

Telephones at different sites must be configured on different subnets or assigned from different address ranges so that the ShoreTel system can properly assign the voice switch for the IP telephone site.

9.7 Configuring DHCP for ShoreTel IP Phones

The ShoreTel server provides the IP phones with the latest application software and the configuration information that enables the IP phone to be automatically added to the ShoreTel system. The ShoreTel server's address must be provided to the phone as a vendor-specific DHCP option. ShorePhones are preconfigured to look for the ShoreTel server's address to be specified as Vendor Specific DHCP option 156. If these options are not available, the ShoreTel IP phones will use option 66.

To set up DHCP option 156 for ShorePhone-IP110/115/212k/230/530/560/560g telephones on a Microsoft DHCP server:

Step 1 Open **DHCP Manager** on your Microsoft DHCP server.

Step 2 Right-click the ***DHCP server***, and select ***Set pre-defined options***.

If your organization is separated into separate subnets, make sure to select the proper subnet. For example, if you have a global organization and would like to configure the DHCP server to deliver the Spanish tones and cadences only to the IP phones in your office in Spain, you should make sure to select that particular subnet of users. If you do not specify the subnet, then all phones that boot from this DHCP server will receive Spanish tones and cadences.

Step 3 Click **Add**.

Step 4 Set **Name** to IP Phone Boot Server.

Step 5 Set **Data Type** to String.

Step 6 Set **Code** to 156 and add a description, if desired.

Step 7 Navigate to the ***scope*** options and add option 156.

Step 8 Set the value of option 156 to:

```
ftpservers=ip_address, country=n, language=n, layer2tagging=n, vlanid=n
```

where ip_address equals the IP address of your ShoreWare Headquarters server.

Refer to Table 9-12 for a list of country codes. Selecting the appropriate country code ensures that the phone has the proper ring tones and cadences needed for a particular country.

Code	Country Name[a]
1	United States of America
2	Canada
3	France
4	Italy
5	Germany
6	Spain
7	United Kingdom
8	Australia
9	Hong Kong
10	Malaysia
11	Singapore
12	Brazil
13	Netherlands
14	New Zealand
15	Portugal
16	Ireland
17	Belgium

Table 9-12 Country codes

a. Check with your system administrator or ShoreTel representative to determine the level of support for a selected country.

Refer to Table 9-13 for a list of language codes. Selecting the appropriate language code ensures that the phone displays the text in the proper language (e.g. abbreviations for days and months, and messages indicating that the phone is requesting service or indicating that service is unavailable).

Code	Language (Country)
1	English (US)
2	Spanish (Spain)
3	German
4	English (UK)
5	French
6	Dutch
7	Spanish (Castilian)

Table 9-13 Language codes

Step 9 Connect the Ethernet cable into the data jack on the back of the IP phone.

The phone downloads the latest bootROM and firmware from the ShoreTel server and in the process, reboots several times. When the phone displays the date and time, the boot and upgrade process is complete.

9.8 Configuring Automatic VLAN Assignment via DHCP

You can configure an IP phone to automatically determine its VLAN id via DHCP. When the phone boots for the first time, it will acquire an IP address via DHCP similar to any other network device. However, the DHCP response will also specify (using a proprietary DHCP option), the VLAN id for the phone to use. Then, the phone will release the IP address originally assigned to it and will reboot. After reboot, all packets are tagged with the VLAN id specified in the original DHCP response.

The following phones are unaffected by this features: AP100 and AP110.

The Automatic VLAN Assignment feature is not configured through ShoreWare Director. Configuration changes are performed at the DHCP server. Parameters related to Automatic VLAN Assignment (along with their supporting text) have been italicized in the procedure that follows to make them easier to spot.

To configure Automatic VLAN Assignment via DHCP on a Microsoft DHCP server:

Step 1 Open ***DHCP Manager*** on your Microsoft DHCP server.

Step 2 Right-click the ***DHCP server*** and select ***Set pre-defined options***.

Step 3 Click ***Add***.

Step 4 Set ***Name*** to IP Phone Boot Server.

Step 5 Set ***Data Type*** to String.

Step 6 Set ***Code*** to 156 and add a description, if desired.

Step 7 Navigate to the ***scope*** options and add option 156.

Step 8 Set the value of option 156 to:

ftpservers=ip address, Layer2Tagging=N, VlanId=X

FtpServers always needs to be set to a ShoreWare server and is a pre-existing parameter.

Layer2Tagging is a new parameter.

Purpose: enable/disable 802.1Q, default is disabled

Format: Layer2Tagging=N

where N=0 is disable, N=1 is enable

VlanId is a new parameter.

Purpose: VLAN id when 802.1Q is enabled, default is zero

Format: VlanId=X

where X is a VLAN id between 0 and 4094

E.g., the following would enable VLAN tagging using a VLAN id of 10:

FtpServers=192.168.0.13,Layer2Tagging=1,VlanId=10

9.9 Time Services

When IP phones are used, time services must be available to maintain the telephone's date and time display. This requires a server that supports the Simple Network Time Protocol (SNTP).

If you do not run an NTP server within your organization, you may use a public accessible time servers used by the NIST Internet Time Service (ITS), shown in Table 9-14.[1]

In addition, you must configure your DHCP server to provide the correct GMT offset to the IP phones at each site. See Section 16.4 on page 229 for more information.

Name	IP Address	Location
time-a.nist.gov	129.6.15.28	NIST, Gaithersburg, Maryland
time-b.nist.gov	129.6.15.29	NIST, Gaithersburg, Maryland
time-a.timefreq.bldrdoc.gov	132.163.4.101	NIST, Boulder, Colorado
time-b.timefreq.bldrdoc.gov	132.163.4.102	NIST, Boulder, Colorado
time-c.timefreq.bldrdoc.gov	132.163.4.103	NIST, Boulder, Colorado
utcnist.colorado.edu	128.138.140.44	University of Colorado, Boulder
time.nist.gov	192.43.244.18	NCAR, Boulder, Colorado
time-nw.nist.gov	131.107.1.10	Microsoft, Redmond, Washington
nist1.symmetricom.com	69.25.96.13	Symmetricom, San Jose, California
nist1-dc.glassey.com	216.200.93.8	Abovenet, Virginia
nist1-ny.glassey.com	208.184.49.9	Abovenet, New York City

Table 9-14 NTP Time Servers

1. This list was obtained at http://www.boulder.nist.gov/timefreq/service/time-servers.html

Name	IP Address	Location
nist1-sj.glassey.com	207.126.98.204	Abovenet, San Jose, California
nist1.aol-ca.truetime.com	207.200.81.113	TrueTime, AOL facility, Sunnyvale, California
nist1.aol-va.truetime.com	64.236.96.53	TrueTime, AOL facility, Virginia

Table 9-14 NTP Time Servers

9.10 Virtual Private Network (VPN)

With the increasing desire to leverage the public Internet, and the concern about security, IP VPNs (Internet Protocol Virtual Private Networks) are becoming the secure access of choice. IP VPNs establish secure communications between employees, branches, or partners by using strong IP-based encryption and authentication techniques for transport security over the public Internet.

IP VPNs are typically viewed as falling into three major categories: remote access VPNs, intranets (company site-to-site), and extranets (business-to-business). These services are being adopted by companies of all sizes as a result of the powerful combination of high-speed access links and public networks. An example is the use of high-speed, low-cost broadband DSL connectivity to enable teleworkers or branch offices to link securely with the company network via the Internet, as if they were accessing the LAN at the office including all network applications. A sample VPN configuration is shown in Figure 9-1.

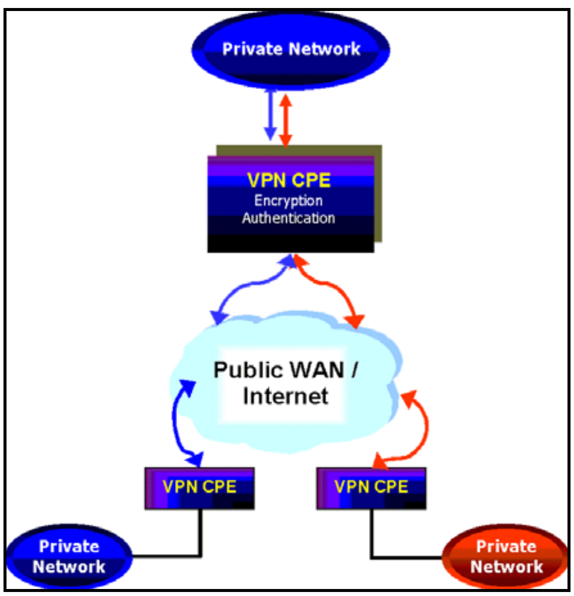

Figure 9-1 VPN Topology

IP VPNs can be provided via hardware or software solutions located at the remote facility (branch office or teleworker's home) and the customer premises. These devices or solutions use technologies such as tunneling, encryption, and authentication to guarantee secure communications across a public infrastructure.

All the components of your ShoreTel system must exist in the same enterprise private network. VPNs can be used to bridge your private networks across the Internet so that the networks for two buildings are both part of the same private network. For multiple locations that share a private network, bandwidth calculations should include the effective bandwidth inside the private network, rather than the raw bandwidth.

9.10.1 Tunneling

Tunneling encapsulates one type of data packet into the packet of another protocol. Multiple tunneling protocols are used today on the market:

PPTP (Point-to-Point Tunneling Protocol): PPTP includes compression and encryption techniques. This protocol was introduced by Microsoft to support secure dial-up access for its desktop, which corresponds to a large share of the desktop market.

L2F (Layer 2 Forwarding): Introduced by Cisco Systems, L2F was primarily used to tunnel traffic between two Cisco routers. It also allows IPX traffic to tunnel over an IP WAN.

L2TP (Layer 2 Tunneling Protocol): L2TP is an extension the PPP (Point-to-Point Protocol) that merges the best features of L2F and PPTP. L2TP is an emerging IETF (Internet Engineering Task Force) standard.

IPSEC: This is a collection of security protocols from the Security Working Group of the IETF. It provides ESP (Encapsulating Security Payload), AH (Authentication Header), and IKE (Key Exchange Protocol) support. This protocol, mature but still technically in a draft format, is currently considered the standard for encryption and tunneling support in VPNs.

For PPTP, IP VPN tunneling adds another dimension to the tunneling. Before encapsulation takes place, the packets are encrypted so that the data is unreadable to outsiders. Once the encapsulated packets reach their destination, the encapsulation headers are separated, and packets are decrypted and returned to their original format.

The L2TP tunneling protocol does not encrypt before encapsulation. It requires the IPSEC protocol to take the encapsulated packet and encrypt it before sending it over the Internet.

9.10.2 Encryption

See Section 9.12 on page 131 for more information about ShoreTel's proprietary media encryption methods.

Encryption is the marking, transforming, and reformatting of messages to protect them from disclosure and maintain confidentiality. The two main considerations with encryption are the algorithm, such as Triple Pass DES (112 bits), RCA (128 bits), and Triple DES (168 bits), and the management of the distribution of encryption keys (IKE and PKI). These more recent keys, which support more than 100 bits, have been a major driver in the success of IP VPNs. They make it extremely difficult to hack into enterprise computer systems without an investment of millions of dollars in equipment.

Encryption starts with a key exchange that must be conducted securely. The IKE (ISAKMP/Oakley) protocol has been considered the most robust and secure key exchange protocol in the industry to date. It is also a de facto standard for service providers and product vendors requiring the highest level of security for their VPN solutions. PKI (Public Key Infrastructure), new to the key management scene, is currently thought to be the long-term solution to simplifying the management of VPNs. The industry is still evaluating and testing PKI, with some initial deployments beginning to occur.

9.10.3 Performance

From an IP VPN[1] performance perspective, encryption can be a CPU-intensive operation. As a result, enterprises must evaluate VPN products in two primary areas as they relate to encryption. The first is whether the maximum throughput decreases substantially when encryption is used, and the second is whether a consistent throughput can be maintained when encryption is enabled. Typically, the trade-off between performance and price is debated from a software-based versus hardware-based encryption perspective.

9.10.4 Integrated Security Appliances

A number of major vendors provide integrated broadband security appliances to eliminate security concerns. These devices use custom ASICs to deliver wire-speed firewall, Triple DES IPSec VPN, and traffic shaping in an easy-to-deploy, cost-effective solution. Installing a security appliance, such as a NetScreen-5, eliminates the need to deal with complex PC software installations and allows IT to centrally manage the security policies of these remote offices and teleworkers. The firewall protection secures sensitive data at the remote site and can prevent both U-turn attacks and the launching of denial-of-service attacks from these computers. By combining broadband access technologies with an integrated security appliance, enterprises and service providers can safely and securely capitalize on all of the benefits of the broadband Internet.

9.11 Firewalls

A firewall is the first major purchase and the foundation of network security (Figure 9-2). It prevents unauthorized access to the network or web site by examining both incoming and outgoing traffic. Based on the predefined security policies, each individual packet is inspected and processed. Any type of traffic that is deemed to be "illegal" (based on rules that specify protocol type, source or destination IP address, and so on) is not allowed through the firewall. Using this tool, administrators can achieve tight control over the activities they allow into and out of their corporate network or e-business site. In a corporate network, a firewall prevents intruders from accessing corporate resources while allowing employees Internet access. In an e-business site, it allows outside access to the web server while preventing unauthorized access or attacks.

Often, a typical network access point, called a DMZ (demilitarized zone), is implemented to offer an "outside" presence for e-commerce clients, e-business partners, and web surfers. The DMZ acts as the gateway through which all Internet communications with the company or site transpire. It allows for controlled access to front-end web servers while protecting mission-critical resources (databases, routers, servers, and so on). Thus, the DMZ needs to be flexible, reliable, and available.

The firewall is often the first line of defense in this environment. Always vigilant, this device must look into all traffic for the site. As part of its duty, the firewall recognizes and deals with denial-of-service attacks, such as TCP SYN flood and Ping of Death. In each of these attacks, the hackers are simply attempting to overwhelm the devices that provide an Internet presence for the company.

With a TCP SYN flood, a stream of TCP SYN packets is sent to the receiving device (often the firewall). The finite memory and size of the TCP entry tables can be overrun by spurious SYN packets, preventing any real users from making a TCP connection required for HTTP communications.

An ICMP flood attack also floods a device, by streaming ICMP echo packets at a recipient destination. This flood of packets requires the device to process and respond to these pings, burning precious resources and preventing other traffic from being serviced. By examining the site's traffic patterns, advanced firewalls can apply logical rules that prevent the device

---

1. Note that Internet VPNs, though useful for data, may not offer sufficient protection against latency and packet loss for VoIP.

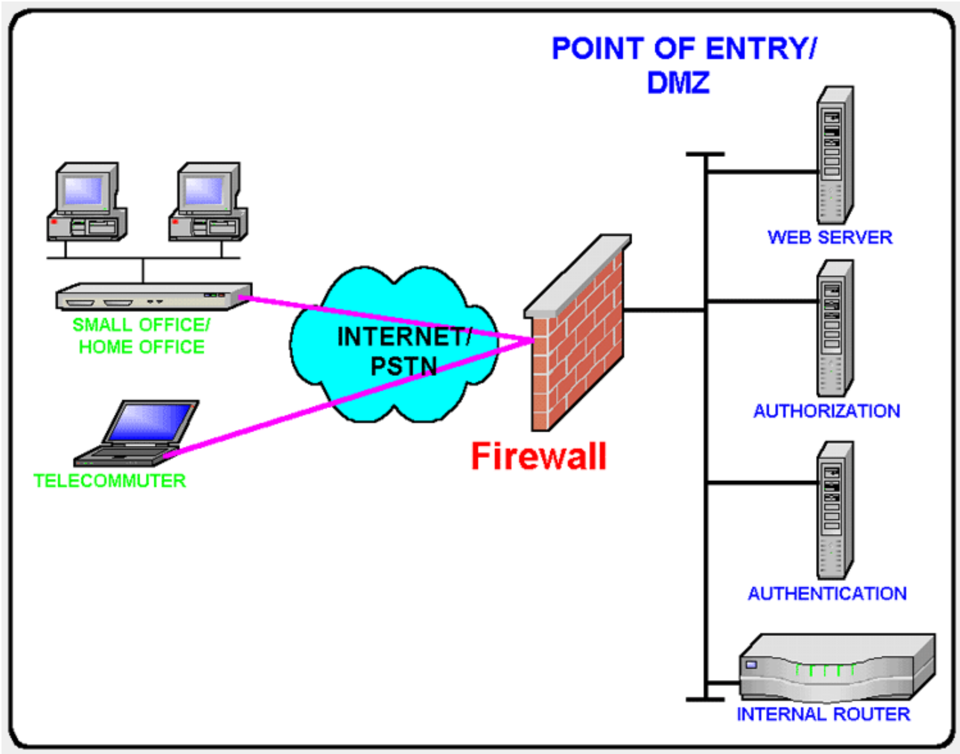

Figure 9-2 Firewalls

from trying to keep up with the denial-of-service attack traffic. They also prevent this traffic from reaching the valuable web, application, and database servers that create your Internet presence and service your customers.

By using firewalls in conjunction with the DMZ design technique, many businesses and service providers are striving to present as much information without permitting unwanted access to the corporate resources.

One way to keep your mission-critical resources as private as possible, while still allowing for a strong Internet presence, is to use Network Address Translation (NAT). NAT offers the outside world one, or a few, IP addresses. This allows a manager to set up whatever internal IP addressing scheme may be required by corporate policies and business needs. An internal resource's IP address (source IP) is changed as it passes through the NAT function to one of the "outside" IP addresses. Thus, the external world does not know any of the enterprise's internal IP addresses. Only the NAT device presents an IP address that is known, and used by external devices. The NAT device keeps track of these conversations and performs the IP address translation as needed.

Extending the private network of the corporate LAN to remote sites via VPN is a proven method of deploying a ShoreTelShoreTel system across multiple sites. All IP telephony endpoints (such as ShoreWare server(s), ShoreGear switches, and IP telephones) should participate in the same private network, with firewalls between ShoreTel equipment and the public Internet. If needed, you can elect to open access to the ShoreWare server(s) to access ShoreWare Director via HTTP, using the same precautions you would when exposing any critical server's web services to the public network.

Configuring firewalls to function correctly with VoIP traffic is very difficult. ShoreTel does not recommend deploying ShoreTel equipment across firewalls.

9.12 Media Encryption

In addition to using a VPN or a firewall, another method of enhancing the security on your network is to enable the ShoreTel media encryption feature. Media encryption, as the name suggests, encrypts calls between users on a ShoreTel system. The encryption scrambles communications between callers so an intruder on the network cannot eavesdrop on the conversation.

The ShoreTel encryption algorithm utilizes dynamically generated keys to encrypt the RTP data for the media stream. The payload inside the RTP packets is encrypted by the sending party, and the transmission is decrypted by the receiving party. The ShoreTel algorithm was selected due to its reliability, simplicity and its efficiency – it places very little burden on the switch's CPU even during maximum loads.

9.12.1 Details:

TCP/IP and UDP packet headers are not encrypted.

Only calls inside a ShoreTel network will be encrypted. Once the call passes through TDM or analog trunks or via SIP, the encryption is stripped away and the conversation is no longer encrypted.

The encryption algorithm handles the key exchange between the sending and receiving parties at the time of call setup. If the call starts off without encryption, and encryption is enabled during the middle of a call, the call will remain unencrypted.

There is no difference in the user experience for encrypted and unencrypted calls. Encryption is essentially transparent, and the user will not know if the call is being encrypted or not.

Encryption is not supported on the SoftSwitch, so calls to voice mail or auto attendant are not encrypted.

9.12.2 Supported Platforms

The media encryption feature is supported on the following hardware.

Platform Type	Model
Switches	• ShoreGear 1U Half-width voice switches
	• ShoreGear 1U Full-width voice switches
IP Phones	• IP110
	• IP115
	• IP210
	• IP212k
	• IP230
	• IP265
	• IP530
	• IP560
	• IP560g
	• IP565g

Table 9-15 Platforms Supporting Media Encryption

For instructions on enabling media encryption, refer to the section on Call Control Options in the *ShoreTel Administration Guide*.

9.13 Session Initiation Protocol (SIP)

There are no special network requirements necessary for deploying SIP. The general system requirements should prove adequate for SIP support. With that in mind, please note the following:

SIP devices are supported behind NAT(Network Address Translation) as long as they are configured statically.

To communicate with a SIP device or service provider providing IP trunks over the Internet, you must be able to pass SIP traffic through your firewall. This requires a SIP application layer gateway – a feature provided by some firewall vendors.

SIP signaling uses UDP port 5060.

When using SIP, the RTP port for the voice media stream is dynamic and the SIP endpoints may not always use the same ports to exchange information (in contrast with ShoreTel's proprietary protocol, which always uses port 5004). Thus, if you are using SIP, you must deselect the "Always Use Port 5004 for RTP" check box on the Call Control Options page in Director so that it is not fixed at 5004.

9.14 Example Network Topologies

9.14.1 Single-Site Implementation

Figure 9-3 is an example of a simple, single-site implementation.

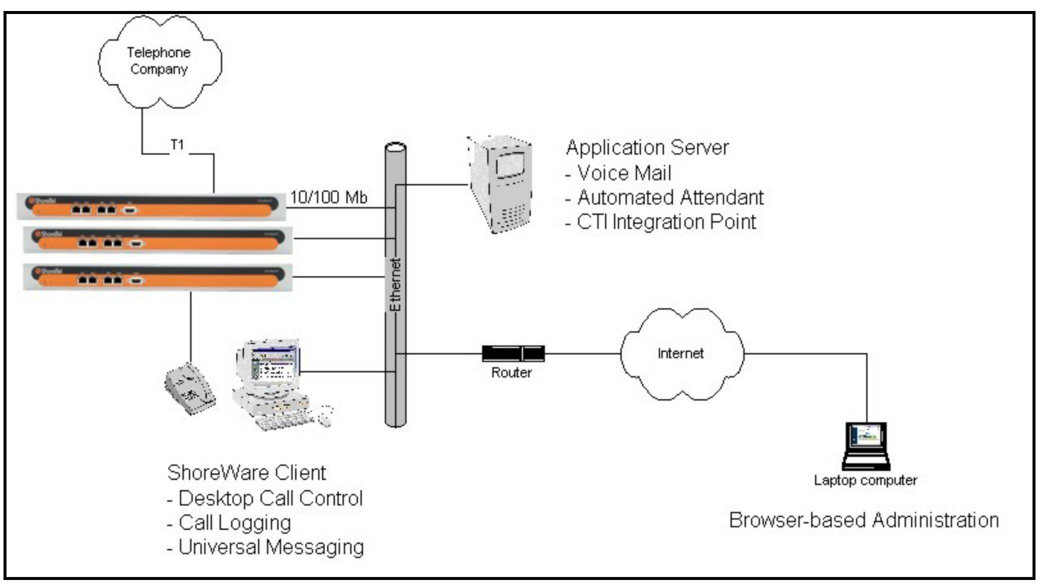

Figure 9-3 Single Site

9.14.2 Multisite Implementation

Figure 9-4 is an example of a multisite implementation with various WAN technology choices.

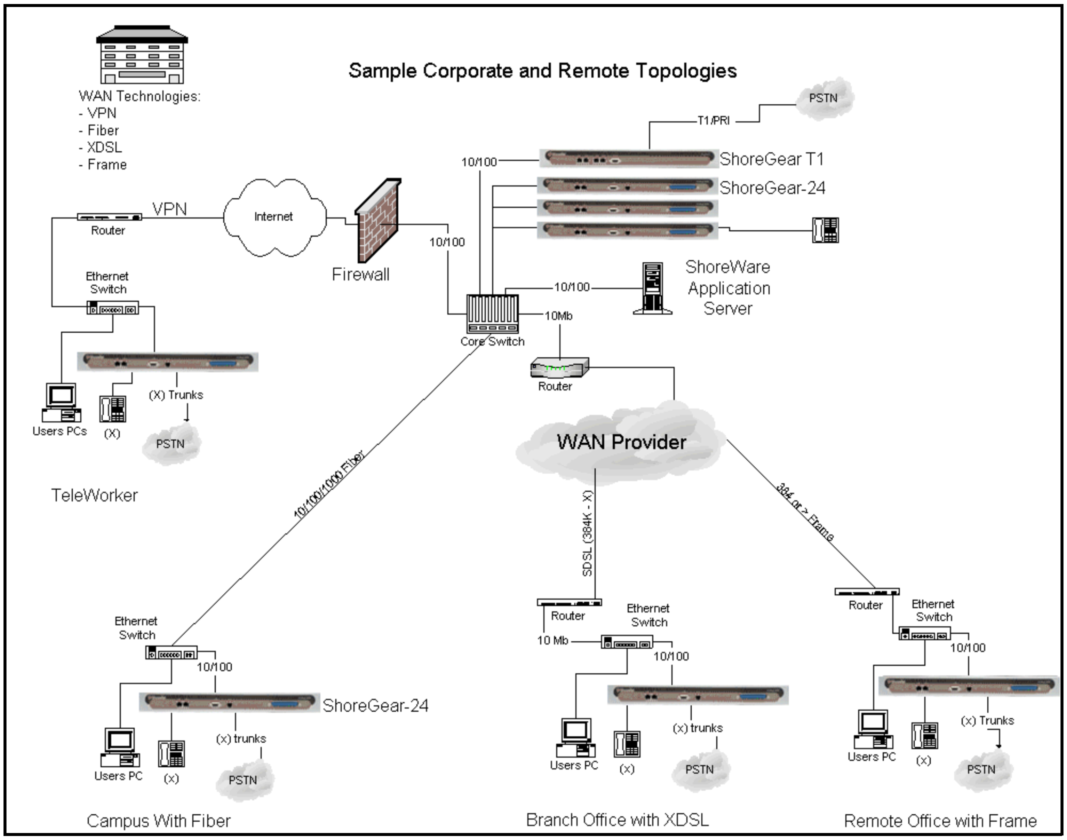

Figure 9-4 MultiSite Options

9.15 Computing Admission Control Bandwidth

This section discusses how to compute the admission control bandwidth for the site you are configuring on the **Site** edit page**—**that is, the appropriate value for the **Admission Control Bandwidth** parameter. If you want to determine the admission control bandwidth for your site and the information is not available in this section, use one of the following formulas:

To determine the admission control bandwidth:

Bandwidth = (# of calls) x (bandwidth/call)

To determine the number of calls supported with a specific admission control bandwidth value:

# of calls = (admission control bandwidth) / (bandwidth/call)

ShoreTel automatically negotiates the proper voice encoder at call setup. For calls between sites, the call control software requests the voice encoder based on what is selected for inter-site voice encoding as defined on the **Call Control Options** edit page. The call control software will then make sure both endpoints on the call can support the requested voice encoder.

For instance, for G.729a voice encoding to be used between two sites, the inter-site voice encoding must be set to G.729a and the ShoreGear voice switches at each end of the call must be G.729a-capable.

9.15.1 WAN Bandwidth per Call (Full Duplex) Without cRTP

Table 9-16 defines the bandwidth, including IP overhead, that is used for each voice call between sites when RTP Header Compression (cRTP) is not being used. The bandwidth depends on the voice encoding used. For example:

> If you want to support 10 calls between this site and all other sites, and G.729a voice encoding is used, set the admission control bandwidth to 260 Kbps. Before you enter this value, make sure the bandwidth is available at this site.
>
> If you set your admission control bandwidth to 768 Kbps and G.729a voice encoding is used, you can support up to 29 calls between this site and all other sites.

ShoreTel recommends that you configure the admission control bandwidth to be less than the bandwidth of the actual WAN link. This provides sufficient bandwidth for call control signaling and other data traffic.

Bandwidth in Kbps per Number of Calls	Linear	G.711	ADPCM	G.729a
1	146	82	52	26
2	292	170	104	52
3	438	255	156	78
4	584	340	208	104
5	730	425	260	130
6	876	510	312	156
7	1022	595	364	182
8	1168	680	416	208
9	1314	765	468	234
10	1460	850	520	260
11	1606	935	572	286
12	1752	1020	624	312
13	1898	1105	676	338
14	2044	1190	728	364
15	2190	1275	780	390
16	2336	1360	832	416
17	2482	1445	884	442
18	2628	1530	936	468
19	2774	1615	988	494
20	2920	1700	1040	520
21	3066	1785	1092	546
22	3212	1870	1144	572
23	3358	1955	1196	598
24	3504	2040	1248	624
25	3650	2125	1300	650
26	3796	2210	1352	676
27	3942	2295	1404	702

Table 9-16 Bandwidth Without cRTP

Bandwidth in Kbps per Number of Calls	Linear	G.711	ADPCM	G.729a
28	4088	2380	1456	728
29	4234	2465	1508	754
30	4380	2550	1560	780

Table 9-16 Bandwidth Without cRTP

9.15.2 WAN Bandwidth per Call (Full Duplex) with cRTP

Some routers support a feature called RTP Header Compression (cRTP) that significantly reduces the amount of IP overhead associated with voice over IP. Table 9-17 defines the bandwidth used between sites when cRTP is being used. For example:

> If you want to support 10 calls between this site and all other sites, and G.729a voice encoding is used, set the admission control bandwidth to 120 Kbps. Before you enter this value, make sure the bandwidth is available at this site.
>
> If you set the admission control bandwidth to 256 Kbps and G.729a voice encoding is used, you can support up to 21 calls between this site and all other sites.

ShoreTel recommends that you configure the admission control bandwidth to be less than the bandwidth of the actual WAN link. This provides sufficient bandwidth for call control signaling and other data traffic.

Bandwidth in Kbps per Number of Calls	Linear	G.711	ADPCM	G.729a
1	132	68	38	12
2	264	136	76	24
3	396	204	114	36
4	528	272	152	48
5	660	340	190	60
6	792	408	228	72
7	924	476	266	84
8	1056	544	304	96
9	1188	612	342	108
10	1320	680	380	120
11	1452	748	418	132
12	1584	816	456	144
13	1716	884	494	156
14	1848	952	532	168
15	1980	1020	570	180
16	2112	1088	608	192
17	2244	1156	646	204
18	2376	1224	684	216
19	2508	1292	722	228
20	2640	1360	760	240
21	2772	1428	798	252

Table 9-17 Bandwidth with cRTP

Bandwidth in Kbps per Number of Calls	Linear	G.711	ADPCM	G.729a
22	2904	1496	836	264
23	3036	1564	874	276
24	3168	1632	912	288
25	3300	1700	950	300
26	3432	1768	988	312
27	3564	1836	1026	324
28	3696	1904	1064	336
29	3828	1972	1102	348
30	3960	2040	1140	360

Table 9-17 Bandwidth with cRTP

There are two ways to set admission control:

Determine the expected number of simultaneous intra-site calls for a site, and multiply this number by the bandwidth required for each call for your selected inter-site encoding.

When admission control is set this way, calls routing between sites will be blocked if placing the call would exceed the number of calls supported by the configured bandwidth.

For information about ShoreTel's Admission Control feature, see Section 9.4.8 on page 119.

9.15.3 Setting Admission Control

The Admission Control Bandwidth parameters are set in the Site edit page of ShoreWare Director. For information on setting this parameter, see Chapter 3, "Configuring Sites" in the *ShoreTel Administration Guide*.

CHAPTER 10

Server Requirements

The information in this chapter helps you determine the specific hardware and software requirements for your main and distributed ShoreWare servers.

10.1 Checklist

Review the following server requirement topics before proceeding to the next chapter:

	Task	Description
❑	Hardware Requirements	page 138
❑	Hard Disk Space Utilization	page 138
❑	Software Requirements	page 139
❑	Software Installation	page 139
❑	Additional Considerations	page 141

Table 10-1 Server Requirements Checklist

10.2 Recommendations

The following recommendations will assist you in procuring and installing your ShoreWare server:

Use a dedicated server for the ShoreWare server. The ShoreWare server provides voice mail, automated attendant, workgroups, and call detail recording, as well as desktop call control services. These are all business-critical applications that should run on a dedicated server.

ShoreTel does not support installation on a virtual server (such as VMware) as real-time voice applications such as voicemail may not have adequate system resources.

ShoreTel does not support the ShoreWare server for use as a Domain Controller.

Select a server from a reputable manufacturer. Servers from clone manufacturers are not recommended for business-critical applications.

ShoreWare Call Quality Monitor (CQM) is software that diagnoses network issues that may result in VoIP stability problems. Call Quality Monitor can be installed on the same server as ShoreWare Headquarters or Distributed Server.

— When installing the CQM on the same device as a Headquarters Server, the ODBC drivers supplied with CQM should not be installed.

— When installing the CQM on the same device as a Distributed Server, installation of the ODBC drivers supplied with CQM is required to assure proper CQM operation.

10.3 Hardware Requirements

For ShoreTel 11.1 Server Hardware requirements and specifications, refer to Chapter 2 - System Overview in this guide and the ShoreTel 11.1 - ShoreWare Release Notes.

10.4 Hard Disk Space Utilization

Approximately 1600 MB of hard disk space is used on the server for program software. Additional hard disk space is used for voice mail, call detail records (main server only), and log files.

Type	Space Required
ShoreWare Server	1600 MB
ShoreWare Remote Server	800 MB
ShoreWare Client	600 MB[a]

Table 10-2 Hard Disk Space Requirements

a. This amount may be necessary when installing off the network due to the installer also being copied.

10.4.1 Voice Mail

Each user's voice mail messages are stored on his or her respective server. The hard disk space used on each server for voice mail varies depending on the number of users, the number of messages per user, and the duration of each message.

You need approximately 30 MB of hard disk space per hour for voice mail storage.

Table 10-3 provides some conservative guidelines to estimate the amount of hard disk space used for voice mail, assuming each user has 15 one-minute voice messages.

# Users	# Messages	Length (minutes)	Storage (hours)	Storage (GB)
100	15	1	25	0.8 GB
500	15	1	125	3.8 GB
1,000	15	1	250	7.5 GB
2,000	15	1	500	15.0 GB
3,000	15	1	750	22.5 GB
4,000	15	1	1,000	30.0 GB
5,000	15	1	1,250	37.5 GB

Table 10-3 Voice Mail Hard Disk Space

10.4.2 Call Detail Records

For each call on the system, call detail records are generated on the main server. The hard disk space used on the server for call detail records varies depending on the call load on the system. The amount of hard disk space for a typical system is shown in Table 10-4.

# Calls/Day	# Calls/Month (20 days[a])	Storage/Month	Storage/ 3 Months
100	2,000	3 MB	9 MB

Table 10-4 Call Detail Records

# Calls/Day	# Calls/Month (20 days[a])	Storage/Month	Storage/ 3 Months
1,000	20,000	30 MB	90 MB
10,000	200,000	300 MB	900 MB
50,000	100,0000	1,500 MB	4,500 MB

Table 10-4 Call Detail Records

a. 20 working days per month (i.e. 4 weeks/month * 5 days/week = 20)

10.4.3 Log Files

Log files are generated on the system for the purposes of technical support. The hard disk space used on the server for log files varies greatly, depending on the overall system activity.

The size of the log files on the server is controlled by parameters within ShoreWare Director. Log files are stored between 1 and 30 days (default 7 days) with a size limit between 0.5 GB and 5 GB (default 4 GB).

File Size	Storage (GB)
Minimum	0.5 GB
Default	4.0 GB
Maximum	30.0 GB

Table 10-5 Log File Hard Disk Space

10.5 Software Requirements

ShoreWare Main and Distributed Server software is tested and certified on the following platforms:

- Windows Server 2003 SP2 (Standard, Enterprise) - 32 bit
- Windows Server 2003 R2 (Standard or Enterprise) - 32 bit
- Windows Server 2008 SP2 (Standard or Enterprise) - 32 bit
- Windows Server 2008 R2 (Standard or Enterprise) - 64 bit
- VMWare Vsphere 4.0

10.6 Software Installation

10.6.1 Installing Microsoft Windows Server 2003 Components

This section describes how to install the Microsoft Windows Server 2003 components.

Step 1 Install the Microsoft Windows Server 2003 components:

Under Application Server, enable the **Internet Information Services (IIS)** option, including the following IIS sub-options:

— File Transfer Protocol (FTP) Service
— SMTP Service
— World Wide Web Service, including the following
  ❋ Active Server Pages

- Internet Data Connector
- Server Side Includes
- World Wide Web Service

Unselect **FrontPage Server Extensions**.

FrontPage Server Extensions are installed by default. This option should be disabled because these extensions have been a source of security problems for servers. There are several exploits using these extensions that allow a hacker to gain access to the file system.

Step 2 If you are using RDP and using Windows 2003, you must ensure the following:

There are no Remote Desktop sessions with Options set for Remote Computer sound set to 'Bring to this Computer'.

They must be configured to 'leave at remote computer'.

10.6.2 Configuring DEP Settings Prior to Installing ShoreWare

We strongly recommended that prior to installing the ShoreWare software, you should configure the Data Execution Prevention (DEP) settings such that DEP is only enabled for essential Windows programs and services. The following procedure configures this setting:

Step 1 Click **Start** and then select **Run**.

Step 2 In the command prompt, type **sysdm.cpl**

Step 3 In the "System Properties" window, click the **Advanced** tab to display the following:

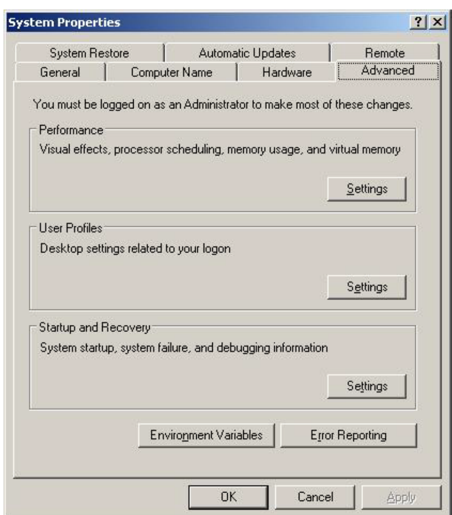

Figure 10-1 Click the Advanced tab

Step 4 In the Performance section, click the **Settings** button.

Step 5 In the window that appears, click the **Data Execution Prevention** tab to display the window shown in Figure 10-2:

Figure 10-2 Configuring DEP Performance Options

Step 6 There are two radio buttons. If it is not already selected, click the upper radio button - **Turn on DEP for essential Windows programs and services only** to select the Opt-In policy.

Step 7 Click **OK** two times to confirm your selection.

10.7 Additional Considerations

10.7.1 Terminal Services

With Windows Server 2003, Microsoft has included the ability to remotely access a server with Remote Administration. In Windows 2003, set up Remote Administration by right-clicking My Computer and then clicking the Remote tab.

Remote Administration allows you to remotely administer a server across the network. In particular, this allows you to launch a terminal session against the main and distributed servers for the purposes of software installation.

ShoreTel also supports Citrix and Terminal Services for ShoreTel client applications. For more information, see Appendix E, starting on page 277.

10.7.2 Adobe Acrobat Reader

Install Adobe Acrobat Reader on the server if you do not already have it, so that you can access the online documentation. You can install Adobe Acrobat Reader from the ShoreWare DVD Browser or download it from the Adobe web site.

10.7.3 DHCP on the ShoreWare Server

ShoreTel does not recommend that the ShoreWare server be used as a Dynamic Host Configuration Protocol (DHCP) server. If you want to use the ShoreWare server to give out IP addresses to the ShoreGear voice switches, you should use the BOOTP server included within ShoreWare Director.

10.7.4 Server Computer Name

You cannot change the computer name of the ShoreWare server after installation. The ShoreWare server software uses Microsoft Transaction Server (MTS), whose license package relies on the name of the computer. Not only will the ShoreWare server not start properly, but you will break the package security if you change the name of the computer.

10.7.5 Server IP Address

The ShoreWare server should have a static IP address to eliminate the possibility that the server will inadvertently get a new IP address, thus adversely affecting system operation.

10.7.6 Internet Information Server (IIS) Default Web Site

The web site for ShoreWare Director is <server_name>/ShoreWareDirector. You should not change the default IIS web site of the server to redirect to ShoreWare Director, since this will cause navigation problems within ShoreWare Director.

10.7.7 Access to the Distributed Server Maintenance Page

If you are using Microsoft Internet Explorer 6 and the distributed server is configured with an IP address rather than a server name, you must enable session cookies on your client computer to access the Distributed Server Maintenance Page.

In Internet Explorer, choose ***Tools*** > ***Options*** > ***Privacy*** tab > ***Advanced*** > ***Override automatic cookies — Always allow session cookies***.

10.7.8 Network Connection Before Installation

The server should be connected to the Ethernet network prior to installing the ShoreWare software to ensure the installation properly recognizes the correct interface.

10.7.9 Workgroup Mode

The server should be configured as a "Workgroup" rather than as part of the domain to avoid having group policies impact the DCOM / COM permissions. It is possible to place the server on the domain after installation of the server software.

10.7.10 Microsoft Updates on the Server

ShoreTel performs weekly updates on test systems with the latest Microsoft server and desktop patches. When releasing a new build, ShoreTel publishes Build Notes that lists the Microsoft patches that are certified against the build. ShoreTel also highlights software changes required by the MS patches.

The conservative approach is to turn off regular MS updates until you review the detailed certification provided with each release.

10.7.11 Virus Protection on the Main and Distributed Servers

ShoreTel allows the use of industry standard virus protection software on the main and distributed servers.

NOTE: **The folders and sub folders MUST be excluded from Virus checker software or disk backup/restore software.**

c:\Shoreline Data\temp
c:\Shoreline Data\Database\ShoreTelCDR
c:\Shoreline Data\Database\ShoreTelConfig
c:\Shoreline Data\Call Records 2\Data

WARNING: If the folders listed above are not excluded before installation, your installation of ShoreWare 10 will fail and your system will rollback to the previous version of ShoreWare. This will also result in a corrupted database if you perform nightly backups.

10.8 Installing Microsoft Windows Server 2008 Components

ShoreWare servers require IIS, COM+, SMTP, and FTP. Other required tasks include changing the SMTP and FTP startup type to automatic.

The following sections describe the selections required at pivotal steps in the installation process. These steps must be completed as a prerequisite to installing or upgrading the ShoreWare components discussed in this section.

NOTE: Windows Server 2008 must be activated through Microsoft before installing ShoreWare Server and Remote Server.

10.8.1 Application Server Role

Figure 10-4 displays the required roles for the Application Server for ShoreTel. Selecting IIS and COM+ ensures the installation of these services.

Step 1 Click the Server Manager icon on the Task bar or, select *Start -> Programs -> Administrative Tools -> Server Manager.*

Step 2 In Server Manager right-click ***Roles*** and select ***Add Roles*** on the pop-up menu.

Note: You can also click the *Add Roles* icon on the right pane

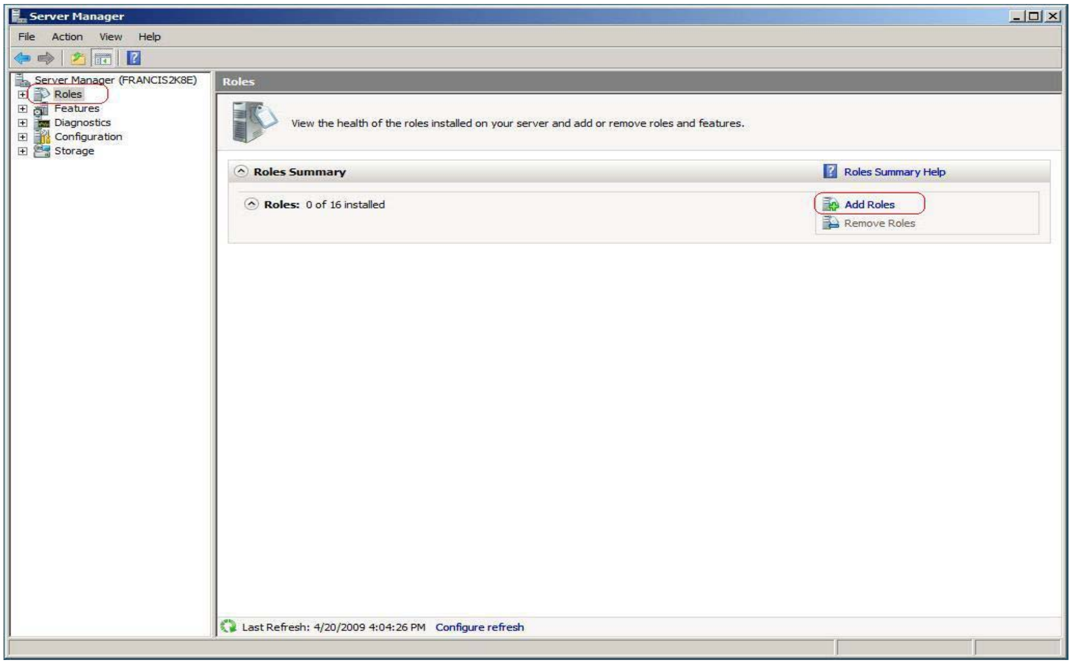

Figure 10-3 Installation Windows Role Services

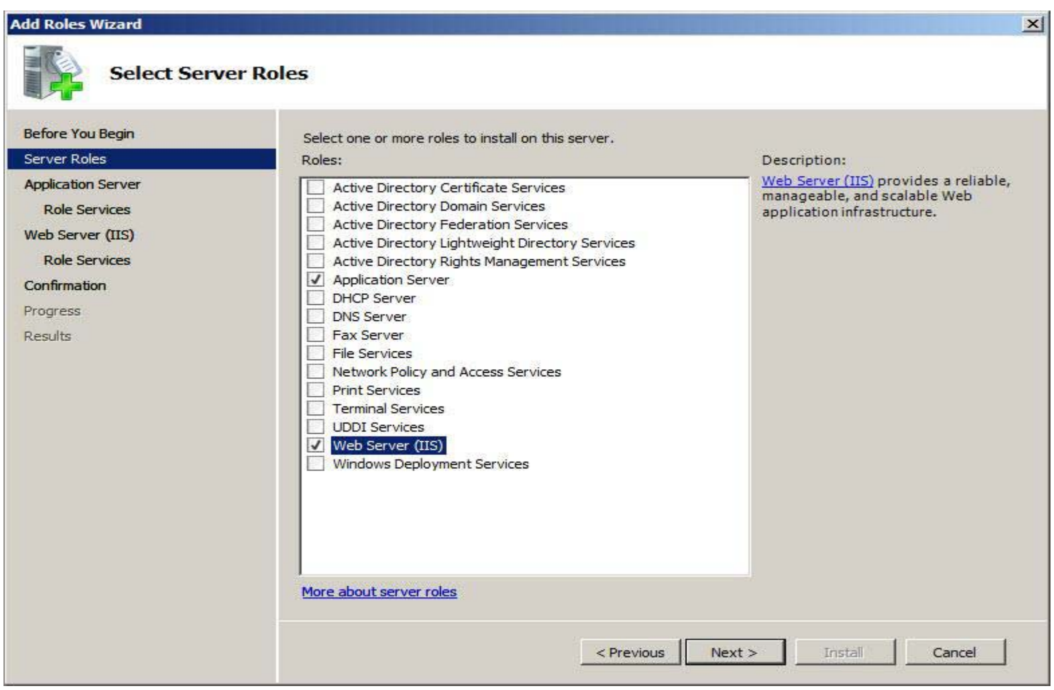

Figure 10-4 Required Server Roles

Step 3 In Add Roles Wizard, click *Server Roles* on the left pane.

Step 4 In the Roles pane, click *Application Server* and *Web Server (IIS)*.

10.8.1.1 Web Server (IIS) Role Servers

Figure 10-5 displays the Web Server required roles.

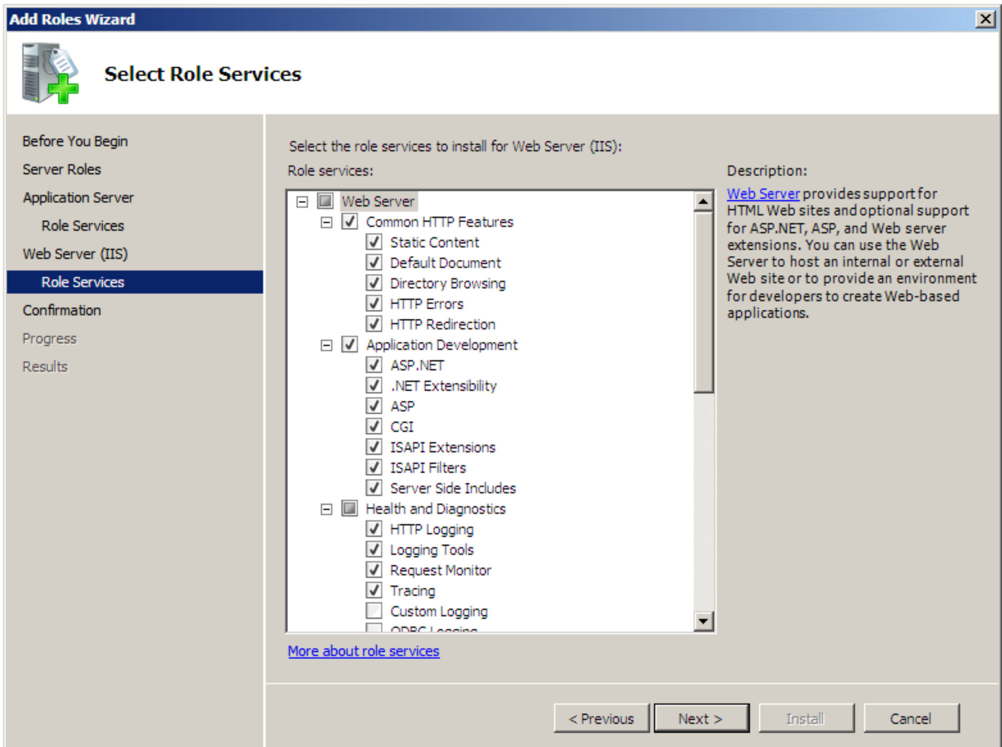

Figure 10-5 Required Web Server Roles

Step 1 Access the Web Server Role by selecting *Server Manager*

Step 2 Right click on *Server->Add Roles*

Step 3 Select *Web Server (IIS)-> Role Services*

10.8.1.2 Web Server Roles

Select all Common HTTP Features

Select all Application Development Features

Select the following Health and Diagnostics Features

- HTTP Logging
- Logging Tools
- Request Monitor
- Tracing

Select all Security options

Select all Performance options

NOTE: You may need to scroll down the Web Role Server window to see all the available options.

10.8.1.3 Management Tools

Select all Management Tools options

10.8.1.4 FTP Publishing Service

Select all FTP Publishing Service options

10.8.2 Microsoft Server Features

ShoreWare requires the installation of SMTP Server. Figure 10-6 displays the Select Features Installation panel.

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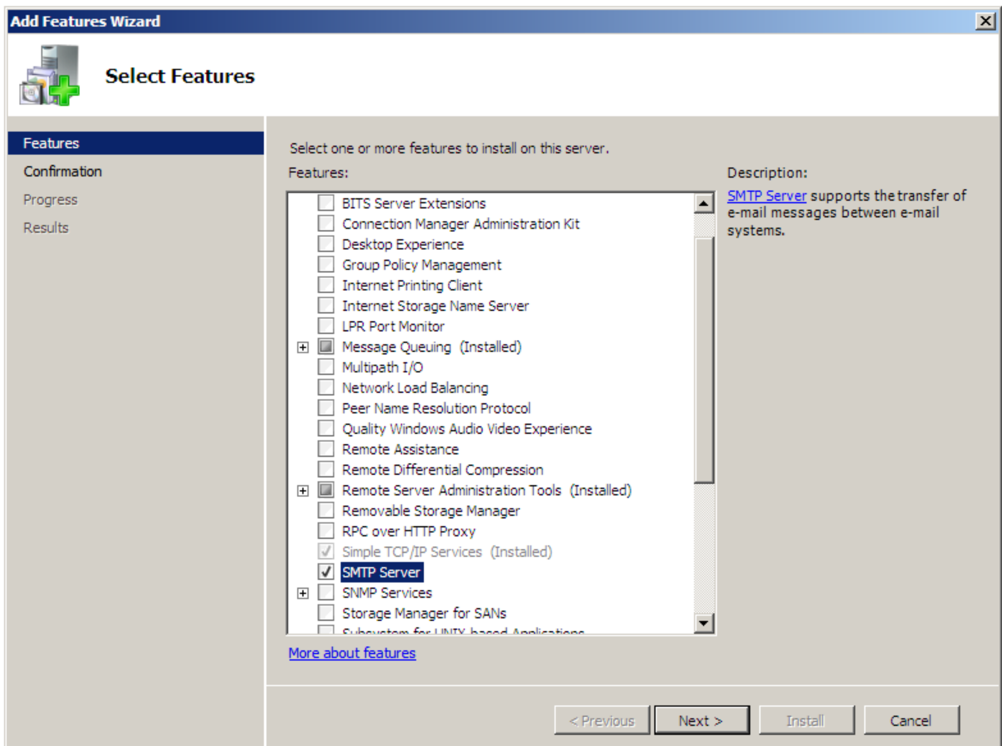


Figure 10-6 Select Features Installation Panel

10.8.2.1 Microsoft Server Feature Properties

After SMTP and FTP are installed, the startup type must be changed from manual to automatic. The following procedure changes the startup type for SMTP and FTP.

NOTE: **Verify that the FTProot folder in the Inetpubs directory has at least read access.**

Step 1 Access the Services table by selecting *Server Manager -> Configuration -> Services*.

Step 2 Open the Simple Mail Transfer Protocol (SMTP) Properties panel by right clicking *Simple Mail Transfer Protocol* and selecting **Properties** on the context menu.

.

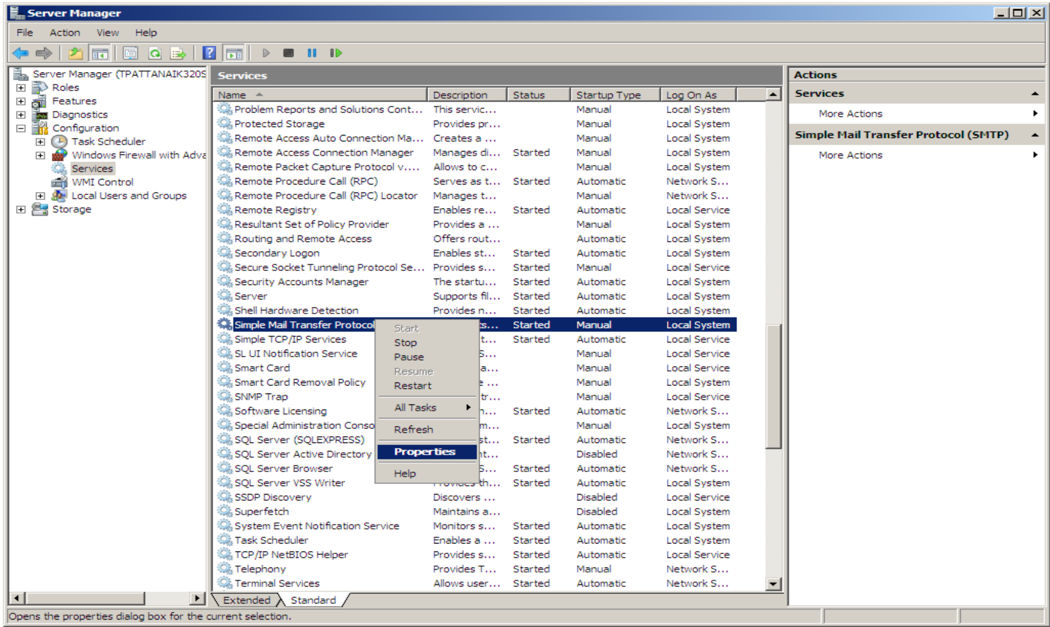


Figure 10-7 Selecting SMTP Properties

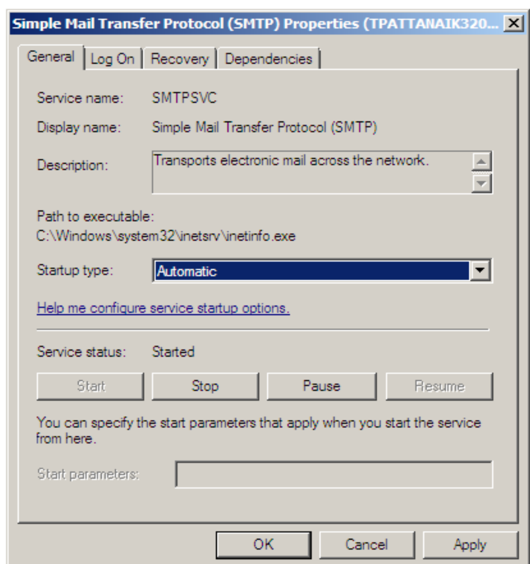


Figure 10-8 SMTP Properties panel

Step 3 Select **Automatic** on the *Startup Type* drop down menu, then click the **OK** button.

Step 4 Return to the Services table.

Step 5 Open the **FTP Publishing Service** properties panel by right clicking ***FTP Publishing*** and selecting **Properties** on the context menu.

Step 6 Select **Automatic** on the *Startup Type* drop down menu, then click the **OK** button.

CHAPTER 11

Installing ShoreTel Voice Switches

This chapter provides planning and installation information for the ShoreTel voice switches. Information on switch connectors and LEDs can be found in Chapter 14, starting on page 191.

11.1 Checklist

Review the following topics before proceeding to the next chapter:

Task	Description
Planning	page 149
Mounting the ShoreTel Voice Switches	page 149
Installing a Voice Switches	page 150
RJ-21X Cable Retainer Installation	page 151
ShoreWare Director Switch Configuration	page 151

Table 11-1 Installing ShoreTel Voice Switches Checklist

11.2 Planning

The requirements to install a ShoreTel voice switch are basically the same as any multisite installation. Please refer to the previous chapters in this guide for more information.

In summary, you must ensure that:

The IP network between the main and remote site meets the bandwidth, latency, jitter, and packet loss requirements for a multisite installation.

The IP network between the main and remote site has quality of service in place such that voice travels ahead of data.

You have appropriate firewall considerations in place, including VPN if applicable.

11.3 Mounting the ShoreTel Voice Switches

To stack the ShoreTel voice switch in a rack:

Step 1 Remove the voice switch from its shipping container.

Step 2 Place the switch on a flat platform, such as a shelf.

Step 3 Up to three switches can be stacked on top of each other.

To mount a full-width ShoreTel voice switch in a rack with brackets:

Step 1 Remove the voice switch from its shipping container.

Step 2 Attach the two mounting brackets, included with the ShoreTel voice switch, using the provided screws.

Step 3 Use standard screws to mount the switch in the rack.

To mount a half-width ShoreTel voice switch in a rack with brackets:

Step 1 Remove the voice switch from its shipping container.

Step 2 Mount a ShoreTel Dual Tray into the rack with the screws provided.

Step 3 Install the half-width switch into the tray on either the left or right side of the tray. Two half-width switches can be placed in the same tray.

Step 4 Use standard screws to mount the switch in the tray.

Refer to the *Quick Install Guide for the ShoreTel Dual Tray* (included with half-width switches) for details.

11.4 Installing a Voice Switches

The DHCP/BOOTP server must be configured prior to turning on the ShoreTel voice switch.

To install a ShoreTel voice switch:

Step 1 Connect the switch to the appropriate LAN segment (such as a LAN switch) with the Category 5 RJ-45 interface cable.

For guaranteed voice quality, all ShoreTel voice switches can be connected to an isolated LAN segment.

Step 2 Plug an AC surge protector into a grounded AC power source (not provided).

Electrical surges, typically lightning transients, are very destructive to equipment connected to AC power sources.

Step 3 Plug the power cord into the power receptacle on the switch's back panel, and into an available socket on the AC surge protector. Most ShoreTel switch models do not have a power switch and power on as soon as you connect the switch to power.

The power LED flashes momentarily, and remains lit. If the LED is not lit, ensure that the power cord is plugged into the switch and the power source. If the LED continues flashing, there is an internal error. Unplug the switch to power it off, then power it back on. Refer to "Configuring Switches" in the ***ShoreTel Administration Guide*** for a description of the flash patterns and their meaning, or contact ShoreTel Customer Operations at: http://www.ShoreTel.com

Once network communications are established, the network LEDs will indicate that the switch is connected to a 10 Mbps or 100 Mbps Ethernet environment, and that the switch is receiving and transmitting data.

Step 4 If applicable, connect the music-on-hold source to the audio input port.

Step 5 If applicable, connect your site's paging system to the audio output port.

Step 6 Refer to the ***ShoreTel Administration Guide*** to configure the ShoreTel voice switch according to your site's requirements.

Step 7 Connect your trunk and telephone lines using the appropriate connector for your switch. Refer to Appendix G, starting on page 303 for connector pinout information for each switch.

11.4.1 RJ-21X Cable Retainer Installation

A cable retainer for the RJ-21X port is included with some ShoreTel voice switches. The retainer consists of a metal bracket with a velcro strap.

To install the retainer:

Step 1 Using a number 1 Phillips screwdriver, remove the two black Phillips head screws on either side of the RJ-21X port.

Step 2 Place the retainer in the recessed area around the RJ-21X port.

Step 3 Reinstall the two screws.

Step 4 Plug in the RJ-21X cable.

Step 5 Pull the velcro strap tightly around the connector on the RJ-21X cable, and fasten it.

11.5 ShoreWare Director Switch Configuration

To complete the installation, you need to configure the ShoreWare voice switches with ShoreWare Director. For more information, see the Configuring Switches chapter in the ***ShoreTel Administration Guide***.

11.6 Reference

11.6.1 Environmental Requirements

The ShoreTel voice switches require that the environmental specifications provided in Table 11-2 be met

Parameter	Specification
Operating temperature	0° C to 50° C
Operating humidity (non-condensing)	0% to 90%
Storage temperature	–30° C to 70° C
Storage humidity (non-condensing)	20% to 95%

Table 11-2 ShoreTel Voice Switch Environmental Specifications

11.6.2 Packaging Requirements

Table 11-3 lists the packaging requirements for the following ShoreTel voice switches:

Full-width switches – (ST-120/24, ST-60/12, ST-40/8, ST-T1/E1)
Half-width switches – (ST-90, ST-50, ST-220T1/E1/T1A)

Parameter	Specification
Vibration	
Power:	0.4 Grms, 1h per axis
Spectral Density:	5-500Hz @ 0.000323303 g2/Hz
Operation	
Power:	1.5G RMS
Spectral Density:	5-500Hz @ 0.00454645 g2/Hz
Packaged Transportation	
Material:	275 C Brown
Dimensions (full-width switches):	21+1/8 x 19 x 5+3/4
Dimensions (half-width switches):	19+1/8 x 12+1/2 x 6+1/2
Mechanical Shock:	80 Gs non-operating
Packaged Bounce:	8-corner standard drop test

Table 11-3 ShoreTel Voice Switch Packaging Specifications

11.6.3 Regulatory Compliance

Parameter	ShoreTel-24
Safety	UL 60950, 3rd Edition, CAN/CSA 22.2 No. 60950, EN60950 (2000)
EMI	FCC Part 15, ICES-003, EN 55022, Class A/Class B
	Radio and Telecommunications Terminating Device Directive (R&TTE) 99/5/EC
	Low Voltage Directive 73 / 23 / EEC
	EMC Directive 89 / 336 / EEC With Amendment 93 / 68 / EEC
	GS Mark from TUV Rheinland (Notified Body)
	EN 55024 : 1998 +A1:2001 +A2:2003

Table 11-4 ST-E1 Voice Switch Physical Specifications

Parameter	Physical Specification
Safety	UL 60950, 3rd Edition, CAN/CSA 22.2 No. 60950, EN60950 (2000)
Telephony Registration	FCC Part 68, Canada CS-03
EMI	FCC Part 15, ICES-003, EN 55022, Class A
	Radio and Telecommunications Terminating Device Directive (R&TTE) 99/5/EC

Table 11-5 ST 120/24, ST 90, ST 60/12, ST 50, ST 40/8, ST 30Voice Switch Physical Specifications

Parameter	Physical Specification
	Low Voltage Directive 73 / 23 / EEC
	EMC Directive 89 / 336 / EEC With Amendment 93 / 68 / EEC
	GS Mark from TUV Rheinland (Notified Body)
	EN 55024 : 1998 +A1:2001 +A2:2003

Table 11-5 ST 120/24, ST 90, ST 60/12, ST 50, ST 40/8, ST 30Voice Switch Physical Specifications

11.6.4 Physical Specifications

Parameter	Physical Specification
Safety	UL 60950, 3rd Edition, CAN/CSA 22.2 No. 60950, EN60950 (2000)
Telephony Registration	FCC Part 68, Canada CS-03
EMI	FCC Part 15, ICES-003, EN 55022, Class A
	Radio and Telecommunications Terminating Device Directive (R&TTE) 99/5/EC
	Low Voltage Directive 73 / 23 / EEC
	EMC Directive 89 / 336 / EEC With Amendment 93 / 68 / EEC

Table 11-6 ST T1, ST-220T1, ST 220T1A, ST T1k Voice Switch Physical Specifications

11.6.5 General Specifications

Parameter	ShoreTel-24
Power Supply	100-240 VAC 50-60 Hz 2A max (full-width switches) 1A max (half-width switches)
Mounting Options	19 inch rack mount
Integrated OA&M	

Table 11-7 ST-120/24, ST-90, ST-60/12, ST-50, ST-40/8, ST-E1/T1, and ST-220T1/E1/T1A Voice Switch Specifications

CHAPTER 12

Planning Applications and Services

This chapter reviews the key applications and services of the ShoreTel system to assist you in planning your system configuration, and to determine the equipment you need for completing deployment.

12.1 Checklist

Review the following application planning topics before proceeding to the next chapter:

Task	Description
Account Code Collection Service	page 155
Voice Mail	page 156
Planning Fax Handling	page 161
Private Numbers	page 169
Automated Attendant	page 169
Call Handling Delegation	page 170
Web Access	page 170
Bridged Call Appearances	page 170
Hunt Groups	page 171
Pickup Groups	page 174
Workgroups	page 175
ShoreTel Communicator	page 177
Enterprise Telephony Features	page 178
ShoreGear Converged Conference Bridge	page 181
ShoreTel Contact Center Solution	page 182

Table 12-1 Planning Applications and Services Checklist

12.2 Account Code Collection Service

ShoreTel supports account codes for external calls when you enable the Account Code Collection Service. When a user dials a number that is not included in the scope of his or her call permissions, the call is routed to the Account Code Collection Service extension, where the user is prompted to enter a valid account code. Account code collection is enabled on a per-user group basis and can be set to be one of three states: disabled, optional, or forced. The Account Code Collection Service is associated with a configurable extension and has a dedicated user group that defines ultimate call permissions and trunk group access.

A new user group is created during installation for use by the Account Code Collection Service. This user group is named “Account Codes Service.” Since it is only intended for use by the Account Code Collection Service, this 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.

The Account Code Collection 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 ShoreGear switch, the call is 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 are automatically placed, and calls placed by users who have forced account code collection are automatically rejected.

12.2.1 Account Codes

Account Code Collection Service supports up to 50,000 account codes of a maximum of 20 characters. You can include non-numeric characters (such as hyphens and slashes) in the account codes; however, non-numeric characters are not used in account code collection or in the account code reports. An account code can be the same as a prefix for another account code. For example, the account codes 1234 and 12345 can coexist.

The following table gives example account codes and how the Account Code Collection Service interprets the code.

Sample Account Code	Recorded Code
Sales 200	200
1001-3	10013
1.234A	1234
3000 Exec 2	30002

Table 12-2 Account Code Interpretation Example

Account codes can also have user-friendly names of up to 50 characters.

12.2.2 Call Permissions

The call permissions define what dialed numbers are 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 is completed with the trunk access and call permissions of the Account Codes Service.

This structure imposes two sets of permissions on outbound calls:

The call permissions for the user group of the user who places the call are used to determine if an account code must be collected or not.

The call permissions for the Account Codes Service determine whether calls are finally placed, or if the intercept tone is to be played.

12.3 Voice Mail

The ShoreTel system provides voice mail for all users and workgroups on the system. The system supports up to 21 application servers—one main server and up to 20 distributed servers. Any of the servers can host the voice mail application.

You should provision a distributed server at any site with more than 100 users to effectively manage your WAN bandwidth between that site and the headquarters or main site. In addition, you must add a distributed server with the voice mail application at any site where the required number of mailboxes exceeds 1,000.

Users should be configured for the server that is located at their home or most frequent site. If that site does not have a server, the nearest server or headquarters server should be used.

When there are multiple voice mail servers, the system-wide voice mail extension automatically maps to the extension of the local voice mail server. Voice mail media streams are therefore recorded in the CDR reports by the voice mail extension that actually handles the call.

The ShoreTel system provides each user with five call handling modes, and workgroups with four call handling modes, allowing employees and workgroups to customize how calls are routed. Employees typically use Standard call handling mode to route calls to voice mail after three or four rings, and use Out of the Office call handling mode to route calls directly to voice mail.

Users should consider:

- Forwarding calls to a cell phone
- Forwarding calls to an external answering service (for critical users or workgroups)

You must enable external call handling as part of the class of service for users who want to use these options.

The Message Notification feature of the ShoreTel system allows users to be notified when they receive a message. Notifications can be sent upon receipt of all messages, or only upon receipt of urgent messages. Notifications can be sent to:

- An E-mail address (with or without the voice mail attached as a .wav file)
- A pager (which allows message notification)
- An extension (which allows message playback)
- An external number, such as a cell phone (which allows message playback)
- Users who address and compose voice mail through the Telephone User Interface (TUI), the Visual Voicemail application, or the Outlook Voicemail form can now mark composed messages for a “return receipt.”

12.3.1 Escalation Notifications

Similarly, the ShoreTel system can send any of these notifications types to specific members of an escalation profile, in support of an Escalation Notification feature. The Escalation Notifications feature is a traditional voice mail feature that allows support groups to offer round-the-clock service to their customers, such that when a customer calls into the ShoreTel system and leaves a message, the voice mail system sends out a page, phone call, or email to a designated employee in the support department. If this first employee ignores his beeping pager, the next designated employee within the escalation profile list is contacted, and so on. Employees in the escalation profile will continue to be contacted sequentially until someone listens to the voice mail. (See “Configuring Users” in the ***ShoreTel Administration Guide*** for more information on this feature.)

12.3.2 Auto-deletion of Voice Mail Messages

The ShoreTel system also supports the ability to automatically delete user voicemail messages that are older than a specified time limit. The system administrator can set a maximum time limit for the storage of voice mail messages, and if this time limit is exceeded, messages are automatically deleted. The tool can be used to encourage users to better manage their voice mailboxes.

12.3.3 Mailbox Full Notifications

The ShoreTel system can be configured to notify users when their voice mailboxes are almost full. This features warns users of the impending lack of storage space to give them ample time to delete messages, as opposed to logging into their voice mailbox only to discover that the mailbox is full. Once a user's mailbox has passed a threshold, the system sends a notice informing them that their mailbox is almost full and that there is only enough room for 10 additional messages. Thusly, users are not caught off-guard by an unexpected (and unwanted) "mailbox full" notification.

For more information, see the "Configuring Users" chapter in the ***ShoreTel Administration Guide***.

12.3.4 Distributed Voice Mail

ShoreTel has Distributed Voice Mail to provide greater availability. Each ShoreWare Remote Server has an instance of the telephony platform, allowing full functionality of voice mail and auto-attendant services at that location during WAN outages. The Distributed Voice Mail feature allows users with mailboxes on that server to receive and pickup voice mail messages without having to depend on a WAN connection to the headquarters server that hosts the configuration database. The message waiting indicator (MWI) lights correctly update local users about voice mail with or without WAN connectivity.

Additionally, incoming calls reach the auto-attendant, access the dial-by-name directory, and reach their intended local party during a WAN outage. If a party cannot be reached directly and his call handling setting would send unanswered calls to voice mail, the call is handled by the local voice mail server. If the user's voice mailbox resides on a different voice mail server, the local ShoreTel server will accept, store and forward the message when connectivity to the proper voice mail server is restarted. The caller hears a generic greeting including the intended party's recorded name and the caller has the option to leave a message. This message will be forwarded at a later time to the home voice mail server for the addressee via SMTP.

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

The ShoreTel Communicator applications may provide limited call control access and may not display some contents on IP phones at a remote site during WAN outages. These both require connectivity to the headquarters server for full service. For users who have their ShoreTel Communicator application running at the time of a WAN outage, graphical access to their voice mail box is provided, including the ability to compose and playback messages, but ShoreTel Communicator may not display the corresponding call activity associated with any actions.

The enhanced Distributed Voice Mail services bring a new level of availability to existing remote servers and allow additional deployment of remote servers up to a system total of 20 remote servers.

12.3.5 AMIS Protocol Support

The ShoreTel system can send and receive voice mail messages to and from legacy voice mail systems using the AMIS protocol Version 1 - Specification; February 1992. To send voice mail messages to remote AMIS sites, ShoreTel dials the access phone number for the remote system. Likewise, to receive voice messages from a remote system, the remote system must know the number to dial into the ShoreTel system. To reach the ShoreTel system, the remote system must be configured to dial any 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 ShoreWare 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.

To simplify AMIS systems, and increase usability:

Use the same extension length across your enterprise.

Use off system extensions to match remote users' mail boxes with their extension numbers.

To identify the remote site location, assign each system a System ID.

For more information on AMIS systems, see the ***ShoreTel Administration Guide***.

12.3.6 SMDI Protocol Support

The ShoreTel system supports the SMDI protocol. Two modes of operation are supported:

In the first mode of operation, the ShoreTel system acts as a PBX for a legacy voice mail system. The ShoreTel system provides call information for forwarded or direct calls to the legacy voice mail system, and receives incoming message waiting indication from the legacy voice mail system.

In the second mode of operation, the ShoreTel system acts as the voice mail system for a number of users on a legacy PBX.

Both configurations require a serial link between a ShoreTel server and the legacy voice mail system, as this is the medium required by the SMDI protocol.

If using the first mode mentioned above, a group of analog trunks must be used to connect the ShoreTel system to the legacy voice mail system (the ShoreTel system is on the extension side of the trunks). The ShoreTel voice mail application manages the group of outgoing extensions. The ShoreTel server can provide digit translation if the legacy voice mail and ShoreTel system have different extension lengths.

It is possible to have some ShoreTel users on the ShoreTel voice mail and some on the legacy voice mail. However, these users will not be able to send messages to each other unless AMIS is implemented between the two systems. Voice mailboxes for workgroups and agents must be on the ShoreTel voice mail system.

ShoreTel Communicator operates the same way it does when a user has no mailbox:

Voice mail viewer is not available

Windows Control Panel does not contain Voice Mail tab

Find Me and Notification features are not available

Dial Mailbox and Transfer to Mailbox are not available for this user from other user's clients

To Voice Mail button on ShoreTel Communicator transfers the call to the system voice mail extension

For more information about using a serial link and SMDI protocol to integrate the ShoreTel system with a legacy voice mail system, see Chapter 15, starting on page 201.

12.3.7 Find Me Call Handling

Find Me and Auto Find Me call handling allow callers to find users at other locations when they reach the user's voice mail. When Find Me is enabled for the current Call Handling Mode, inbound callers that reach a ShoreTel user's voice mail box can activate Find Me call handling by pressing "1." If the caller activates Find Me call handling, the system plays a prompt indicating that it is now finding the called party: "*Please hold while I try to find your party.*"

ShoreTel users can specify two Find Me destinations, which can be internal or external numbers. These numbers can be enabled or disabled for each Call Handling Mode. If a call is forwarded to the first number and is not answered within a configurable number of rings, the call can either be forwarded to a second Find Me destination or can be returned to voice mail.

The Caller ID that appears on Find Me calls is the voice mail Caller ID and not the ID of the original caller. However, if the source of the original call is external to the system, then the Caller ID will be displayed. Personal Assistant (pressing "0") also works when Find Me forwarding is enabled. The voice mail system dials the configured Find Me numbers in sequence. When a Find Me call is answered, voice mail announces the call through a sequence of prompts.

The party that answers a Find Me call hears prompts similar to the following:

"I have a call for Sam Smith from 4085551212."

"To accept this call, press one."

"To send this call to voice mail, press two."

"To repeat the caller ID, press three."

The party at the Find Me number has three options for directing the call:

Pressing **1** connects the original caller with the intended party at the Find Me destination.

Pressing **2** directs the voice mail system to immediately start taking a message for the intended party from the original caller.

Pressing **3** repeats the Caller ID information available on the call, if any. This also extends the timeout by 1 ring (6 seconds).

The voice mail system does not automatically notify callers of the Find Me call handling option. ShoreTel users can elect to tell callers of the Find Me option in their recorded greeting (i.e. they can tell callers to "press 1 to Find Me"). If the user does not tell callers about the Find Me option in their greeting, the Find Me option can remain a hidden capability available only to selected callers. Conversely, users can automate the Find Me behavior so that when a call enters voice mail (and Auto Find Me is enabled), the call is immediately sent to the Find Me destination numbers without requiring any action on the part of the caller.

12.3.8 Call Sender

Users can place a return call to the originator of a voice mail by pressing "5" from the phone during message playback. Users can also call back the voice mail sender from ShoreTel Communicator, Agent Monitor, or Microsoft Outlook, if the user is so provisioned. To use this feature, the user must belong to a user group with trunk-to-trunk transfer Class of Service enabled. For more information, see the ***ShoreTel Administration Guide***.

The user has the option of replying with either a voice message or a phone call if Caller ID information is available on the call. If no Caller ID information is available for the call (for example, on calls from an outside caller), the "reply with a call" option is not available for that message.

When the user chooses to reply with a phone call, the call is transferred to the number of the originating party. When the originating party is an external caller, the message recipient must have the dialing permission to dial the Caller ID number. Once the message recipient is transferred to the number of the message originator, there is no option to return to the mailbox.

12.3.9 Time Stamps

The time stamp of the message is relative to the time on the server where the message is taken. For example:

- When the user views messages in the Voice Mail Viewer or Outlook Form, the user interface will adjust the time stamp based upon the time of the user's computer.
- When the user dials into voice mail to retrieve their messages, the time stamp will be based on the time of the server.

12.4 Planning Fax Handling

The ShoreTel system supports fax calls. There are several ways to configure your fax service.

- A direct fax number for each site
- Direct fax numbers for each user (using either individual fax machines or a fax server)
- Redirect faxes that are sent to the site's main number to a fax machine extension at the site
- Redirect faxes that are sent to a user's extension to user's local fax extension

Figure 12-1 shows how to plan your fax options.

How you configure your fax service with ShoreWare Director depends on which method of fax call handling you have chosen. The following provides a basic outline of the steps involved:

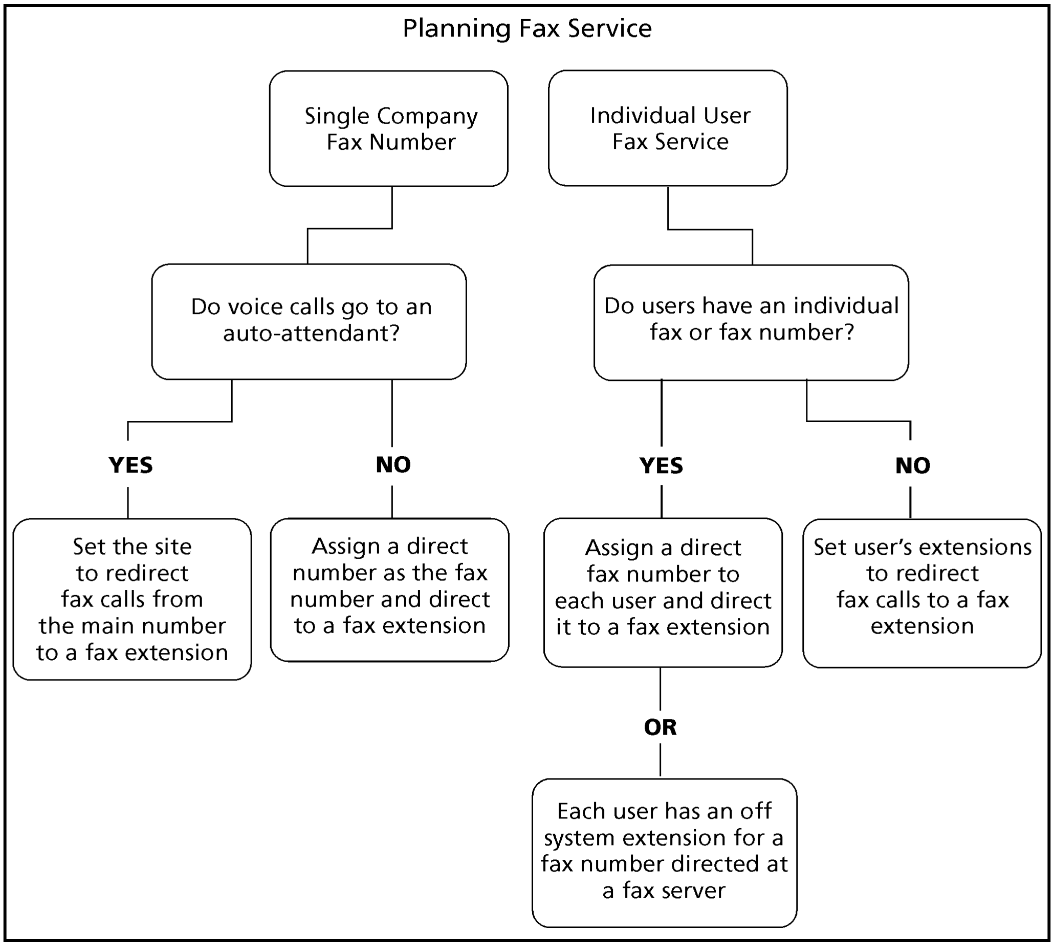


Figure 12-1 Planning Fax Service

If you plan to use the main number for voice and fax calls, and the main number goes to an auto-attendant:

Step 1 Configure the fax extension through the ***User edit*** page of ShoreWare Director.

Make sure that fax redirection is disabled for fax extension "users."

Step 2 Enter a fax extension you created in Step 1 in the FAX Redirect Extension field from the ***Site edit*** page.

If you plan to use the main number for voice and fax calls, and the main number goes to an operator:

Step 1 Configure the fax extension through the ***User edit*** page of ShoreWare Director.

Step 2 Assign a direct number as the fax number.

Step 3 From the ***Trunk Group edit*** page (on the DNIS map page), set the destination to the fax extension.

If your users have their own faxes or fax service:

Step 1 Configure the fax extension(s) through the ***User edit*** page of ShoreWare Director.

Step 2 Assign a range of direct fax numbers.

Step 3 From the ***Trunk Group edit*** page (on the DNIS map page), set the destination for each fax number to the appropriate fax extension.

If you plan for each user to have a single number for both voice and fax:

Step 1 Configure the fax extension(s) through the ***User edit*** page of ShoreWare Director.

Step 2 Enable fax redirection from the ***User edit*** page and enable fax redirect for the site by entering a fax extension on the site edit page.

For more information on these settings, see the ***ShoreTel Administration Guide***.

12.4.1 Using a Fax Server

A fax server improves services available to your users, helping them be more productive. With a fax server, users can:

- Send faxes directly from the desktop eliminating the need to print faxes to send.
- Receive faxes directly on the desktop.
- Integrate fax communications with e-mail and voice mail applications.
- Have individual fax numbers
- Maintain soft copies of all faxes for easy printing and document management.

Using a fax server with the ShoreTel system allows you to:

- Share inbound and outbound trunks for fax services.
- Reduce toll charges by leveraging your VoIP network for outbound faxes.

For inbound fax support, users can be assigned a personal fax number from the DID range of one of the trunk groups and this DID number can be the same as the user's regular telephone extension. When a call is received, if the fax redirect feature is enabled, the system can differentiate between voice calls and fax calls and react appropriately.

Outbound faxes are queued by the server and then sent across the IP network to the best available trunk.

Fax Server Requirements

- Sufficient ports on ShoreGear voice switches
- Sufficient ShoreWare User Licenses
- Sufficient DID trunks to support both fax and voice DID for all users

12.4.1.1 Network Requirements

The network requirements for faxing over IP are more stringent than for voice over IP. For voice communications, a 1% packet loss has negligible impact on voice quality. However, a 1% packet loss for fax communications means a loss of approximately 3 lines per fax page. ShoreTel recommends that packet loss not exceed 0.1% across the LAN and WAN when using fax servers with the ShoreTel system.

Fax communications are also impacted by voice compression. Since fax machine typically require 19.2 Kbps, ShoreTel recommends that you use G.711 voice encoding for fax calls. For more information on fax requirements, see Section 8.4 on page 103.

Note that the fax redirect feature will not work with calls that come in on SIP trunks.

12.4.1.2 Fax Server Integration Details

Instead of requiring users to have two separate DID numbers (one for voice and one for fax) a single DID line can handle voice calls and inbound/outbound faxing.

A user's extension (which can be 3, 4, or 5 digits) is sent to a fax server via in-band Dual Tone Multi Frequency (DTMF) digits. The fax server uses this information to create a mapping between the user's extension and his or her email address.

Once configured, incoming fax calls are received at the user's phone extension. The fax server listens for the fax tone, takes over the call (assuming the fax redirect radio button has been selected in Director). When the fax transmission is complete, the loop current is automatically turned off to terminate the fax call, and the and fax is forwarded to the associated email address.

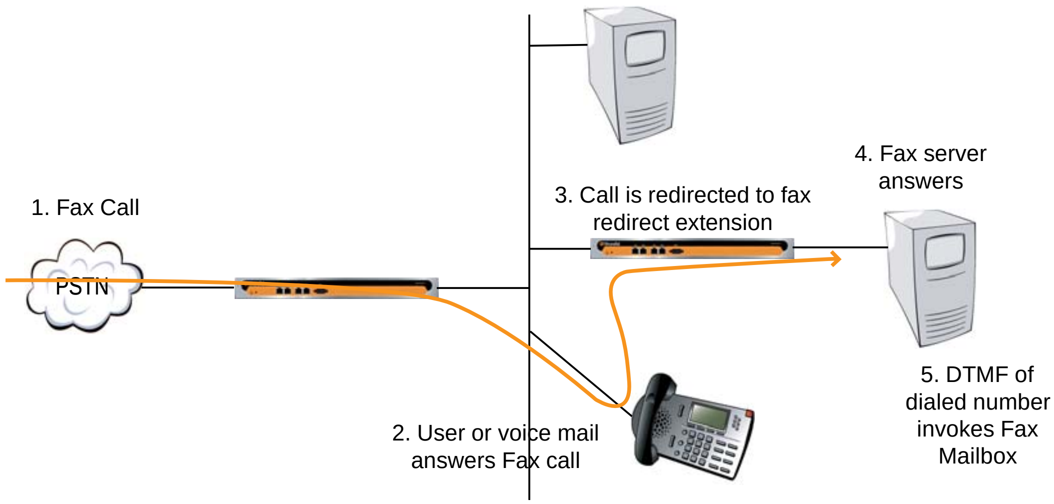


Figure 12-2 Fax server integration call flow

12.4.1.3 Enhanced FAX Server Integration

In addition to calls redirected from a user's extension, the ShoreTel system will now deliver digits to a Fax Server for DID calls routed directly to a FAX server, thus allowing the call to go directly to the fax extension and provide DID/DNIS digits, instead of to an extension number and then to the fax server.

12.4.1.4 Configuring Fax Server Integration

At a high level, the process of setting up the Fax Server Integration feature involves three tasks:

- Connecting the hardware (i.e. connecting the fax server ports to the analog ports on the switch)
- Creating a user account to represent each analog port
- Enabling the Fax Server Integration feature for each user account

To configure the Fax Server Integration feature:

Step 1 Configure a fax server per the manufacturer's instructions.

Step 2 Connect the fax server to one of the **analog** ports on a ShoreGear switch that supports analog. The following switches support fax server integration:

Next, you will create user accounts to represent each analog switch port that connects to the fax server.

Step 3 Launch ShoreWare Director and enter the user ID and password.

Step 4 Click on the ***Administration link*** to expand the list (if it has not already been expanded).

Step 5 Click on the ***Users link*** and then the ***Individual Users*** link, and then ***Add a New User***.

Step 6 The ***Edit User*** window appears, as shown below. (Arrows in the illustration point to fields that must be configured. Refer to the bulleted list below the illustration for details.)

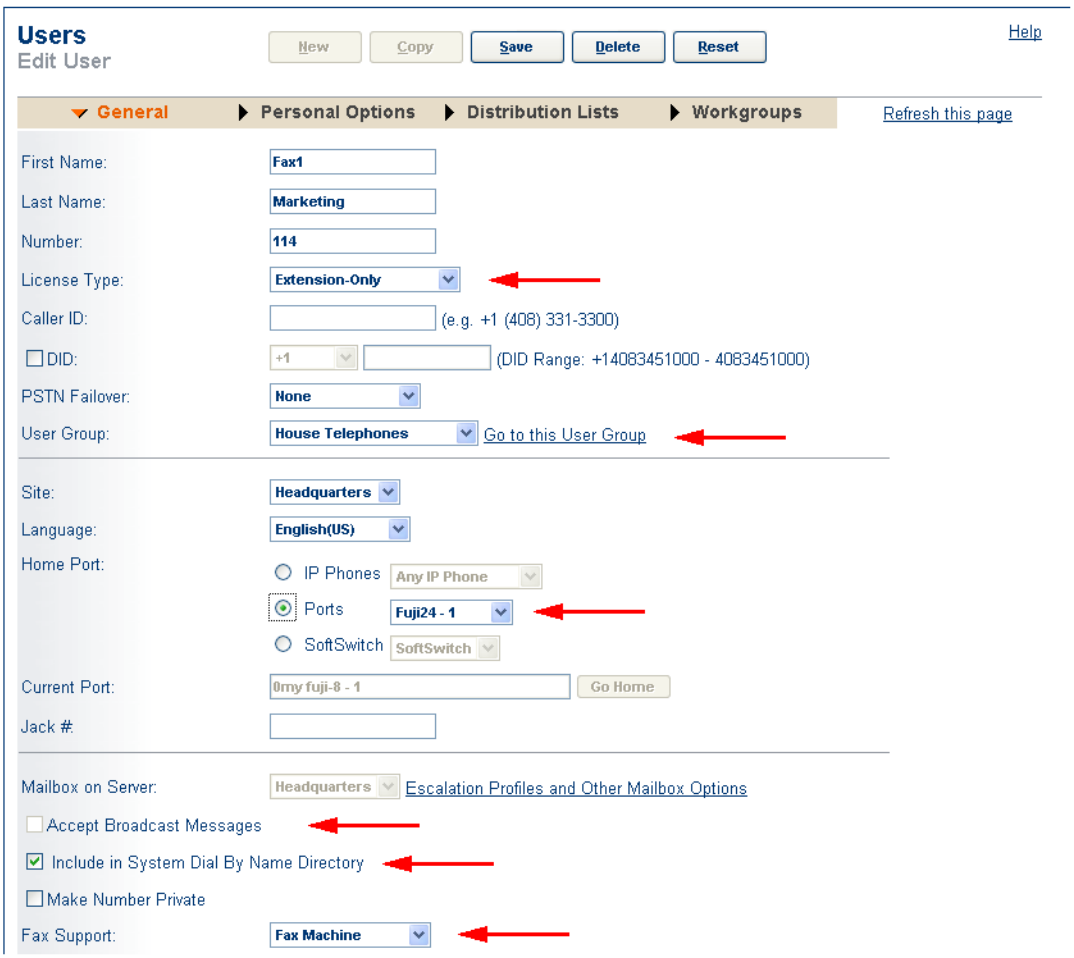


Figure 12-3 Creating a user account for the fax server

Step 7 Enter information for each of the fields as shown below for each field:

License Type: Extension-Only

User Group: You must create a User Group appropriately configured for a fax server. The User Group should have the Class of Service for Call Permissions set to **No Restrictions** to transfer inbound and outbound faxes.

Home Port: Select the **Ports** radio button and then use the drop-down menu to select the switch where the fax server will be connected.

Accept Broadcast Messages: Should appear grayed-out or be deselected because the port will not be assigned a mailbox.

Include in System Dial By Name Directory: Check box may be selected if you want callers to be able to locate the fax number using the Dial by Name feature.

Fax Support: This Extension is Connected to a Fax Server radio button must be selected.

Step 8 Click the **Save** button to store your changes.

Step 9 Click on the **Personal Options** tab and enter "*1*" in the Current call stack size field.

Step 10 Click **Save** to store your changes.

Next, you will configure the call handling mode for each of the user account(s) associated with the port(s) connected to the fax server.

Step 11 From Director, select the user account representing the fax server connection.

Step 12 Click on the **Personal Options** tab.

Step 13 Click on the **Standard** link under Edit Call Handling Modes.

Step 14 Under Call Forward Condition, select the **No Answer/Busy** radio button, as shown below:

Step 15 In the Busy Destination and No Answer Destination radio buttons, select **Extension** and specify the analog port where incoming fax calls will be directed if the first fax port is busy.

For example, if you have set up three ports to receive fax calls, you might configure the first port in this series to redirect to the second port, and the second port would specify the third as a failover.

Step 16 Click **Save** to store your changes.

This configuration assumes multiple analog ports will be used to connect the switch to the fax server. If only one fax server port will be used to connect to the fax server, then the call forwarding must be set to **Never**. Similarly, if this port is the last one in a chain of ports dedicated to the fax server, then the call forwarding must be set to **Never**.

If you are using multiple analog switch ports to connect to the fax server you must specify the first redirect extension in that chain. (This is the site's fax redirect extension.)

Step 17 Under the **Administration** link, click **Sites**.

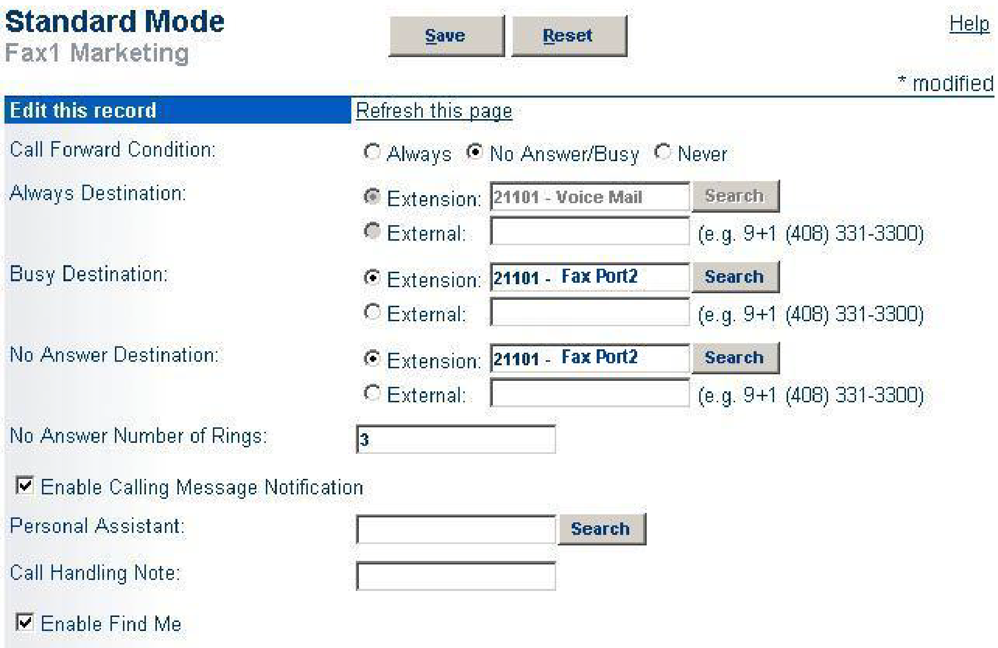


Figure 12-4 Configuring Call handling mode for Busy/No Answer failover

Step 18 Click on the site where the switch and fax server are located (i.e. either Headquarters or Remote).

Step 19 Under **FAX Redirect Extension** (near the bottom of the Site window), enter the extension associated with the first port in the chain of fax server ports. (This is the first place incoming faxes will be sent.)

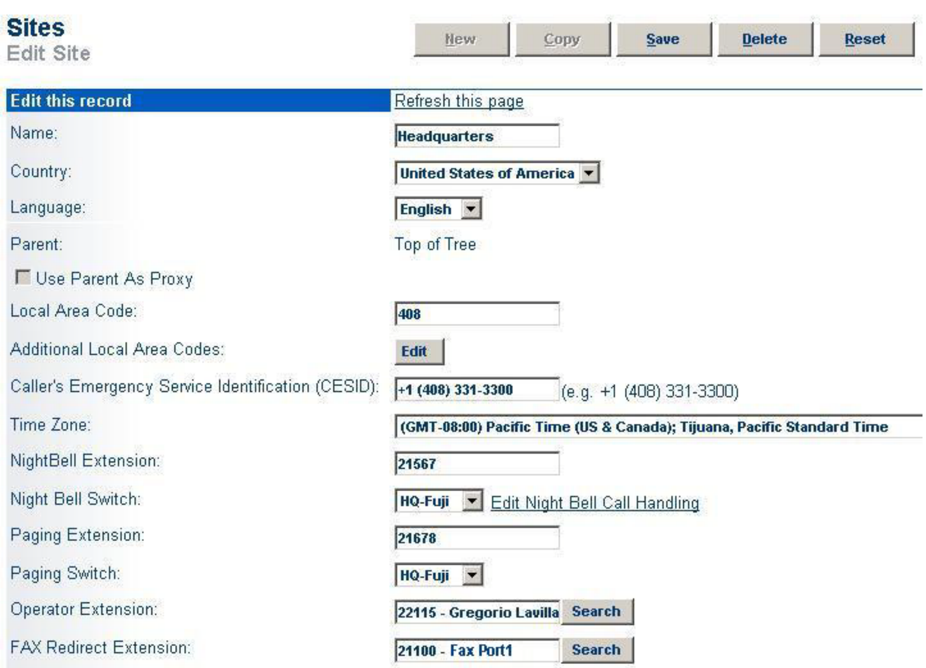


Figure 12-5 Configuring Fax Redirect extension for primary fax server port

Step 20 Click **Save** to store your changes.

Next, you must configure settings for each user that will be using the new Fax Server Integration feature.

Step 21 Click on the ***Users link*** and then the **Individual Users** link.

Step 22 Click on the name of a user who will be using the enhanced Fax Server Integration feature.

Step 23 The ***Edit User*** window appears, similar to the one shown below.

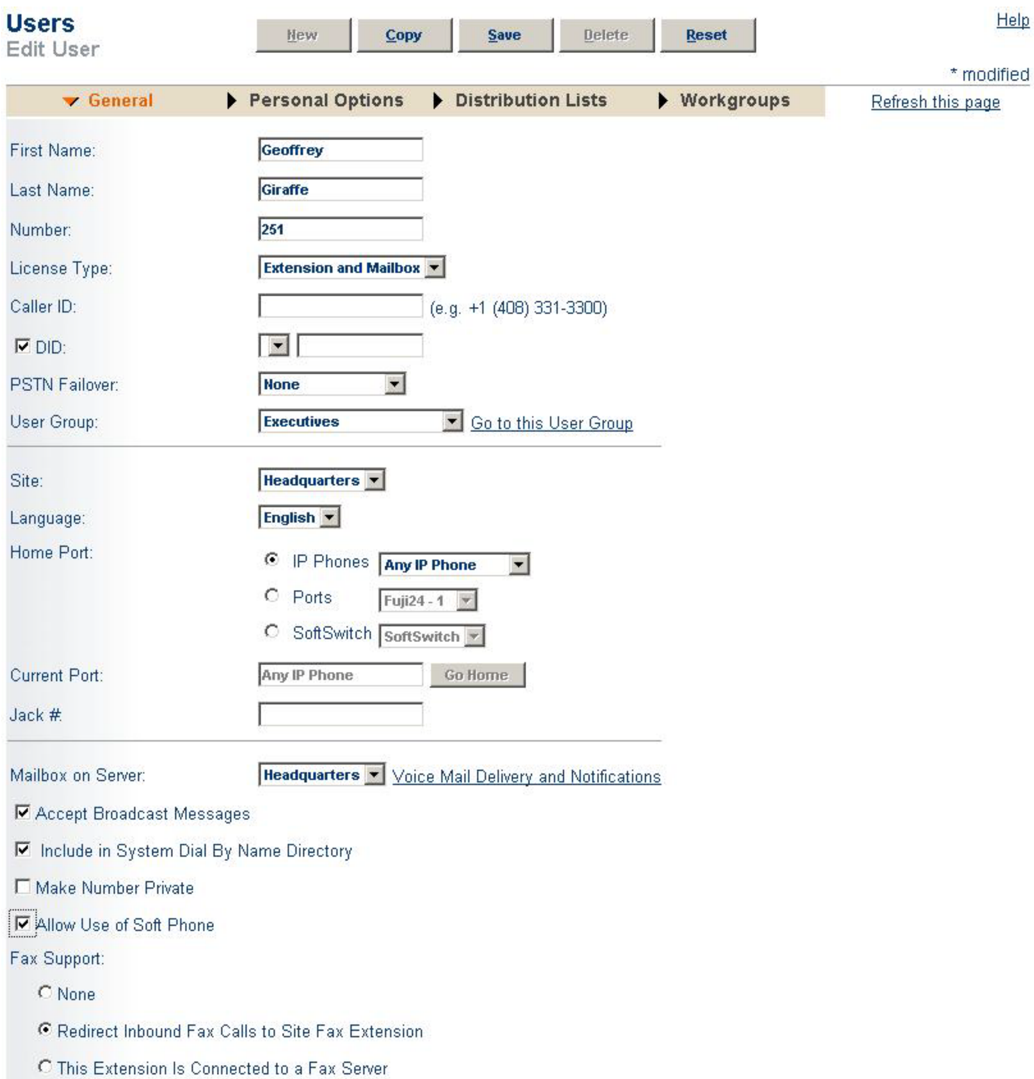


Figure 12-6 Enabling fax redirect for a user

Step 24 Select ***Redirect Inbound Fax Calls to Site Fax Extension*** for the Fax Support radio button.

Step 25 Click the **Save** button to store your changes.

12.5 Private Numbers

Users can have private numbers that are not listed in the System Directory or in ShoreTel Communicator Quick Dialer, and for which Caller ID information suppressed. Private Numbers are enabled through a check box on the User edit page in ShoreWare Director. When checked, the user's extension becomes a Private Number.

The following conditions apply to private numbers:

- Private Numbers do not appear in the QuickDialer for dial-by-name operations or in the ShoreTel Directory Viewer.
- Calls placed from a Private Number to an internal party show the caller's name but not his or her number to the dialed party.
- 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 site CESID number is used for the outbound Caller ID.
- Calls 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 ShoreTel Communicator History viewer show the Private Number user's name but not his or her extension number.
- The Private Number users are listed with name and number in the Extension Monitor extension selection dialog box.
- The Private Number user can be dialed directly via the telephone or the ShoreTel Communicator if his or her extension is known.
- Contacts imported from Outlook or Exchange are never private and are fully visible in the ShoreTel Communicator Quick Dialer.
- CDR database records show both number and name for Private Number users. However, the Caller-ID Flags field indicates that only the name is valid.
- CDR legacy log files show the number of Private Number user calls that are inbound or outbound calls.
- ShoreWare Director shows number information for Private Number users as with other users, for example on the User list page.

12.6 Automated Attendant

The ShoreTel system comes bundled with an automated attendant feature that runs on each of the voice application servers, allowing high availability. The system supports up to 256 menus with four scheduled modes, providing a simple, flexible solution.

Some useful applications for the auto-attendant menus are:

- Answering the main number
- Routing calls to workgroups (sales, support, human resources, and so on)
- Providing automated directions
- Providing a way for users to log in to voice mail (“#” recommended)

Although the automated attendant is a useful tool, you should take care to design a menu structure that does not frustrate your callers. Here are some helpful hints to keep in mind:

- Do not cascade menus more than two or three deep.

Provide a “zero-out” option on every menu, routing the call to a live human being (“0” is recommended).

Remember to provide an option to return to the previous menu (“*” is recommended).

Try to keep prompts short, quick, and efficient.

Users can record AA menu prompts from their own telephone, instead of having to go through Director. This ability frees the system administrator from having to be involved with the task of recording AA menus, allowing him or her to delegate the task to more appropriate team members. For details on enabling this feature, please see the ***ShoreTel Administration Guide***.

12.7 Call Handling Delegation

Some users of the ShoreTel system, particularly senior management, often have an administrative assistant who helps them manage items such as their email, calendar, and voice communication. The ShoreTel system administrator can grant permission from ShoreWare Director to individual users to change another’s current call handling mode (CHM) settings. Users who have been delegated to change CHM settings can make changes to the current CHM settings for other users using Operator ShoreTel Communicator. The Web Access CHM client also includes this capability. For more information on configuring call handling delegation, see the ***ShoreTel Administration Guide***.

12.8 Communicator for Web

Communicator for Web is a browser-based interface that allows users to change their call handling mode and options. Mobile users can change their call handling options from any computer connected to the intranet or Internet. Communicator for Web can be a public URL for remote access or restricted to the LAN.

To open ShoreTel Communicator for Web from within a ShoreTel system:

Step 1 Open your browser and type:

`http://<servername>/client`

in the URL address text box, where `<servername>` is the name of your ShoreTel server.

Step 2 Press *Enter*. The Communicator for Web login page appears in your browser.

Step 3 Log in with you client ID and password.

For information on how to provide Internet access to ShoreTel’s Communicator for Web client using Apache Server as a reverse proxy, see Appenix D.

12.9 Bridged Call Appearances

The Bridged Call Appearances (BCA) feature provides “bridged” information between many separate IP phones, offering the benefit of faster call handling between users. The feature is intended for key system environments, such as a small office with a moderate number of trunks, IP phones and users.

Custom buttons are configured on each IP phone so that information about incoming calls to a BCA extension is shared among the phones via blinking colored LEDs. Similarly, IP phones can share information about outbound calls placed from a BCA extension by blinking green or red on each phone (see the ***ShoreTel Administration Guide*** for details).

Custom buttons can be programmed on IP phone such that each button represents a position in the call stack.

Pressing the top-most BCA custom button for outbound calls does not necessarily access trunk 1. There is no one-to-one correlation between the custom buttons programmed for BCA extensions and a particular trunk. Trunks can be associated with BCA extensions in any random manner desired by the system administrator.

12.9.1 Switch Support for Bridged Call Appearances

The ShoreGear voice switches support BCA functionality, with the following caveats:

Up to 24 BCA extensions can be configured per switch.

The sum of all the trunks that are assigned to a BCA, plus the call stack size of all BCAs used for extension appearances on a switch cannot exceed 24. For example, you may configure 8 BCAs, each targeted with 3 trunks on the same switch.

A maximum of 32 phones can be configured to point to the same BCA extension.

Up to 128 BCA extensions (on other switches) can be monitored.

For details on configuring the BCA feature, please refer to the ***ShoreTel Administration Guide***.

12.10 Hunt Groups

Hunt groups allow you to route calls to a list of extensions. Hunt groups can be accessed through an extension, DID, and/or DNIS. Hunt groups are supported by ShoreGear switches and remain available when connectivity to the ShoreWare servers are lost. The hunt group can be used as the backup destination for a workgroup, so that some basic hunting can be done even when the workgroup server is not reachable. To maximize reliability, assign hunt groups to a switch close to the majority of the members and/or trunks associated with the hunt group.

A maximum of 8 hunt groups can be assigned to a single switch. A total of 16 user numbers can be assigned to hunt groups on a single switch (i.e., 8 hunt groups with 2 extensions each, 2 hunt groups with 8 extensions, or 1 hunt group with 16 extensions).

Hunt groups have scheduled call handling modes similar to route points (for more information about route points, see the “Setting Call Control Options” chapter in the ***ShoreTel Administration Guide***). There are call handling modes for on-hours and off-hours/ holiday (combined). For on-hours, destinations can be set for Always, Busy, and No Answer. For the other call handling modes, only a call forward always destination is provided. When the hunt group is in a call handling mode other than on-hours, the hunt group forwards calls to the Call Forward Always destination.

A hunt group can be a destination anywhere in the system where a workgroup is allowed as a destination. This includes call forward destinations from users, workgroups, route points, personal assistants, site operators, site fax redirect extensions, and Find Me destinations.

12.10.1 Hunt Group Busy State

The hunt group can be set as busy from both the switch maintenance page in Director and with a star code from the Telephone User Interface. This feature allows hunt group members to disable hunt group routing when they are temporarily unavailable or leave work early. The busy state of the hunt group is maintained by the hunt group's switch and is not saved in the configuration database or to flash memory. When a switch boots or reboots, the hunt group is in the "available" state.

Use the star code "*18" followed by the hunt group extension, to toggle the busy state of the hunt group from a telephone. A class of service setting controls whether a user can change the hunt group busy state.

When the hunt group is in the busy state during on-hours, calls are forwarded to the busy destination.

12.10.2 Configurable Hunting

There are two types of hunting available with hunt groups: top down or simultaneous ring. All hunt group members are hunted for each call received. For example, in top-down hunting, if the switch is hunting members for an initial call when a second call is received, the second call hunts through all the members again. In other words, each call is hunted independently and in the case of top down, hunting starts at the top.

You can also configure:

- The number of rings per member (the same number of rings are used for each member to whom the call is offered).
- Whether calls should go to a no answer destination after all members have been hunted once or whether members are rehunted.
- Whether multiple calls are offered to a member simultaneously when the hunt group receives multiple calls. Calls are not offered to members with full call stacks.
- Whether members should be hunted when the member's call handling is set to Call Forward Always (DND).

12.10.3 Hunt Group Applications

Hunt groups provide solutions to a several call routing scenarios.

12.10.3.1 Backup Routing for Workgroup

To use a hunt group as a backup when the workgroup server cannot be reached, create a hunt group with workgroup members who will serve as backup members. To use the hunt group when the workgroup server is not reachable because of a network outage, admission control, or a server outage, set the workgroup's backup number to the hunt group. When the hunt group is set to offer each member a single call at a time, then call offering is similar to a workgroup. Hunt group members are hunted even though they are logged out or in wrap-up with respect to the workgroup.

12.10.3.2 Hunt Group as a Call Forward Destination

In a small office where individuals generally receive calls directly, users may want someone in the office to answer calls when they are unable to answer. To handle this situation, create a hunt group with everyone in the small office as a member. Individual users can set their call forward destinations to this hunt group. The hunt group can be configured with simultaneous ring, to hunt members only once, and to go to voice mail with Call Forward Busy and Call Forward No Answer conditions.

When configured as described above, if a user's call was forwarded to the hunt group after it wasn't answered, the hunt switch hunts everyone in the office. If the call was not answered after the maximum number of rings, the call is forwarded to voice mail where the caller can leave a message in the original target's mailbox.

12.10.3.3 Distribution of Calls to Backup Operators

In this scenario, a primary operator who handles calls to a main company number requires one or more secondary operators to receive the calls when the primary operator becomes too busy.

To create a hunt group to back up the primary operator, create a hunt group with backup operators. Enter the main operator and all the backups as members of the hunt group in the order in which they are to serve as backups. Set the hunt group for multiple calls to be hunted to a given member, and set the call stack size for each of the users to control the number of calls he or she can receive.

When there are incoming calls to the hunt group, the primary operator is offered the calls first. The operator may be offered multiple calls concurrently up to the limit of his or her call stack. If a member's call stack is full, the member is skipped and that particular call is not be offered again (unless the hunt group is set to hunt forever and no member picks up the call before the member is reached again in the hunt list).

If a member of the operator group does not answer the hunt call, the call is offered to the next member after the number of rings configured for call forwarding. Thus, even if the primary operator has room on his or her call stack, the call is offered to the next member in the list when the operator does not answer the call in time.

If you want calls to go directly to a backup when the primary operator is not available, then set the hunt group not to hunt the members when their current call handling mode is set to Call Forward Always (DND). Operators can use this configuration to pass calls to other hunt group members by changing their call handling mode to Call Forward Always.

You may wish to have a hunt group that goes immediately to voice mail or another number during non-working hours. The hunt group can be configured with an off-hours schedule. Setup a schedule for on-hours during which the call handling mode for the hunt group is configured to forward calls to another number only if the hunt group is busy or no one answers. For off-hours, set the hunt group to call forward always to voice mail or another number. The auto-attendant automatically changes the hunt group's current call handling mode based upon the configured schedule.

12.10.3.4 Common Line Monitoring

A hunt group can be used for line monitoring. For example, several operators may wish to monitor the same line and all have an opportunity to answer calls at the same time. For this case, set up a hunt group with simultaneous ring. When a call is received, the hunt switch rings all operators in the hunt group whose call stack is not full to the number of rings configured. If the hunt group is set to hunt forever, when the number of rings is reached the hunt switch rehunts the same users again. However, the members who have room on their call stack for additional calls may have changed, so each additional hunt may result in different phones ringing.

12.11 Pickup Groups

Group Pickup is a traditional PBX and key system feature used in group environments. The feature allows users in a pickup group to answer any ringing phone in that group, and the feature works best in places where several people work together on a daily basis, such as design firms. If a group member is away from her desk and across the room while 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 from an analog phone.

The following example may help illustrate how this feature is used.

Assume three hypothetical users (e.g. Mike, Joe, and Sarah) work together and have jobs that require extensive collaboration. They also sit near one another. Their extensions (x1001, x1002, x1003, respectively) would be added to an extension list, and then this list would be associated with a pickup group.

The pickup group would have its own extension (e.g. x3755). Note that this extension is invalid and cannot be dialed, and thus acts more like a code than an extension. This non-dialable extension could be programmed into a ShoreTel Communicator toolbar button or an IP phone programmable button on Mike's, Joe's, and Sarah's phones.

So, assume Joe's phone rings (x1002) while he is having a conversation with Sarah at her desk. He would hear his phone ringing at his desk, yet he could press the pre-programmed button on Sarah's IP phone in order to answer his own call.

Alternatively, if Sarah had an analog phone, Joe could press *13 + 3755 to answer the call.

Pickup groups can include the following types of extensions:

User extensions
Workgroup extensions
Bridged Call Appearance (BCA) extensions

Details

Pickup groups can be associated with a programmable toolbar button, or with a programmable button on an IP phone, and can work with Extension Assignment.

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 ShoreGear T1 and ShoreGear E1 voice switches.

The call pickup feature will support:

— 24 members per group

— 16 groups per switch

— The sum of all 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.

Users can use this feature in several different ways:

- **IP Phone –** If a programmable button has been configured for Pickup Groups, the user can press the button, or key, and enter the extension for the Pickup Group to answer the call.
- **IP Phone –** If a soft key has been programmed, the user can press the “pickup” soft key and enter the extension to answer the call.
- **ShoreTel Communicator –** If one of the pre-programmed buttons in ShoreTel Communicator has been set up for Pickup Groups, a user can enter the extension of the Pickup 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.
- **ShoreTel Communicator –** Alternatively, the user can access the “pickup” command from the Call Menu, followed by the extension.
- **Analog Phone –** The user can enter the *13 command, followed by the Pickup Group extension to answer calls from an analog phone.

12.12 Workgroups

The ShoreTel system supports up to 256 workgroups, with up to 300 members per workgroup. (The Simultaneous Ring feature is limited to 16 members.) A workgroup enables a group of users to appear as a single unit to calling parties. Calls can be routed in top-down, longest-idle, round-robin, and simultaneous-ring fashion. Workgroups are typically used by support and sales groups to help automate call handling.

The ShoreTel system provides a ShoreTel Communicator - Workgroup Agent Access and ShoreTel Communicator - Workgroup Supervisor Access with the proper software licenses. In addition, you can run workgroup reports on the server to help you understand and assess workgroup activity and performance.

ShoreTel analog phones do not display Caller ID for calls forwarded from a workgroup.

12.12.1 Agent Multiplicity

Users can be members of multiple workgroups. The workgroups can be configured for any hunt pattern and can have queuing enabled.

A single agent status is applied to all workgroups of which the user is a member. With one status, an agent is either logged-in, logged-out, or in wrap-up for all workgroups of which he or she is a member. In order to manage their own logged in status, users must be provisioned with ShoreTel Communicator - Workgroup Agent. Agents can manage their logged-in state via ShoreTel Communicator, or through the TUI menu in their voice mailbox or via their IP phone programmable button.

When an agent is a member of more than one workgroup, that agent can receive calls from any of the workgroups. When an agent is available to take calls from more than one workgroup, and the workgroup would select that agent based on the current hunt pattern for a call, the oldest call is offered to the agent.

Queue Monitor shows calls from all the queues of which the user is a member. If the user is a member of only one queue, there is no change to the interface. However, if the user is a member of multiple workgroups, the Queue Monitor shows statistics for each workgroup, and for all workgroups. The user can specify a filter to show only a subset of the queues. The filter only changes the information displayed and does not alter the hunting behavior; the user will still be offered calls from all workgroups of which the user is a member.

For workgroup supervisors the Agent Monitor shows all agents from the workgroups of which the supervisor is a member. The Agent Monitor also allows supervisors to filter agents being monitored by selecting individual workgroups.

12.12.2 Call Monitor and Barge In

Call Monitor creates a limited conference call where the monitoring party hears the other parties, but the monitored parties do not hear the monitoring party. When a call is being monitored, a warning tone may be played to the participants of the call. The warning tone can be disabled using an option for an Auto-Attendant Menu. Call center administrators typically disable the warning tone to silently evaluate agent performance. When the warning tone is disabled, the menu prompt typically informs the caller that their conversation may be monitored or recorded.

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.

A recording warning tone may be played to the customer during silent monitor. The warning tone is enabled from ShoreWare Director. No tone is played during a Barge In call.

WARNING ShoreTel, Inc. does not warrant or represent that your use of silent monitoring or barge in 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 configuring the call monitoring features, you may wish to consult with legal counsel regarding your intended use.

To simplify discussion of this feature, we will refer to three parties: the supervisor, the agent, and the customer. The supervisor initiates the silent monitor by selecting an agent. The agent is on a call with the customer. The customer may be an internal or external caller, but supervisors and agents must be on extensions.

In Silent Monitor, 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 in a monitored extension, then the silent monitor is dropped.

If a conference call is already in progress, it cannot be monitored. If a silent monitor is already in progress, no one else can monitor the call.

The supervisor can barge in on a call he or she is silent monitoring. It is not possible to revert a barge in call to 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 is the last party added.

12.12.2.1 Barge In and Silent Monitor Telephony COS Configuration

Each telephony class-of-service (COS) permissions has several additional check boxes and radio buttons in ShoreWare Director to configure Intercom/Paging, Barge In, Call Recording, and Silent Monitor.

Allow initiation for Intercom/Paging—If this check box is selected, users within this COS may place an intercom call or page to other system users. If cleared, then no intercom/ paging can be initiated.

Accept Intercom/Paging—Radio button choices are:

Accept None: If selected, users within this COS may not receive intercom calls or pages.

Accept All: If selected, users within this COS may receive intercom calls or pages from anyone in the COS.

Accept Only From: If selected, users within this COS may only receive intercom calls or pages from the person 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 with this COS permission.

Accept Only From: If selected, users within this COS may only receive barge-in's from the person specified in the field accosted 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 of others 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 with this COS permission.

Accept Only From: If selected, users within this COS may only have their calls recorded by the person 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 of others 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 with this COS permission.

Accept Only From: If selected, users within this COS can only be monitored by the person specified in the field associated with this radio button.

There are no special permissions for ShoreTel Contact Center agents or supervisors. They must have these same COS permissions with appropriate settings to enable contact center silent monitoring and barge in.

12.13 ShoreTel Communicator

The ShoreTel system provides a multilevel ShoreTel Communicator to address the various needs of the enterprise user.

For information on ShoreTel Communicator access licenses, see the ***ShoreTel Administration Guide***. See the ***ShoreTel Communicator User Guide*** for details about using Communicator.

12.14 Enterprise Telephony Features

12.14.1 Music on Hold

ShoreTel can provide music on hold on a per-site basis using the audio input port on ShoreGear switches that support music on hold. Refer to Appendix H to determine the switches that support music on hold. You only need a single music source per site.

Connecting the desired music source to the designated ShoreGear voice switch provides music on hold. The source can be either recorded music or custom music, with prerecorded announcements or other information for callers.

Each site with music on hold must have its own music source. To conserve bandwidth, music is not sent across the WAN between sites, and MOH is selected by the ShoreGear Switch where the CO trunks are configured (i.e., the holding party). IP phone users will not receive MOH when they are on an internal call. See the ***ShoreTel Administration Guide*** for additional information.

Before installing the system, confirm that you have music sources for each site, including the music and the required equipment for playback.

Details related to MOH over SIP Trunks:

MOH for SIP trunks is offered for environments where external users reach the ShoreTel system through SIP trunks (such as BRI via a SIP gateway), and MOH will be offered internally, in situations where the SIP protocol is used to reach the ShoreTel system through SIP devices, such as a WiFi phone.

If there is a MOH source at the same site as a SIP trunk, these trunks will be connected to that source when placed on hold, and the device at the other end of the trunk will connect directly to the MOH switch.

The existing rules for MOH will also apply to MOH for SIP Trunks:

- MOH will not be sent across sites.
- The MOH source must be at the same site as the SIP trunk that utilizes it.

Limitations:

MOH is not supported over the SIP tie trunk towards analog phones, analog trunks or PRI trunks. Currently, MOH is sent over the tie trunk and is not generated by the local device.

MOH Works for the following SIP trunk devices:

- Hitachi Phone
- SIP BRI gateway
- PolyCom SIP phone
- SIP Service Provider Network (e.g., Masergy).

MOH is supported across SIP Tie Trunk to IP Phone in the following scenarios:

- From an IP phone to another IP phone
- From an analog phone to an IP phone (i.e. putting the call on hold from an analog phone)
- From any trunk (PRI/analog) while placing an IP phone caller on hold

— From any phone type to a SIP trunk device such as a Hitachi phone over the SIP tie trunk and onto the SIP trunk device

12.14.2 Overhead Paging

The ShoreTel system can provide single-zone overhead paging on a per site basis using the audio output port associated with ShoreGear voice switches that provide an audio output port.

For sites that require overhead paging, you must designate one of the ShoreGear voice switches to provide paging. In addition, you must provision your selected paging equipment for connection to the ShoreTel system.

12.14.3 Paging Groups

As an alternative to a paging system, you can designate groups of system extensions that can be paged by dialing a single system extension. In this way, audio is routed to a group of phones and played on the phone speaker as opposed to playing the audio announcement on an overhead paging system.

With that said, you can also add a paging extension (associated with a site's overhead paging system) to a paging group in order to simultaneously play audio on a group of phones AND an overhead paging system. Refer to the ***ShoreTel Administration Guide*** for details.

Pages to on-hook IP phones will automatically be announced on the IP phone speaker. Pages to IP phones or analog phones that are already on a call will be treated as a normal call. Call handling modes do not apply to page calls.

A maximum of 100 extensions can be paged at one time. Group paging is not available to external callers.

Please refer to Product Bulletin ST0200 on the ShoreCare website for details on setting up Paging Groups and for details on other network considerations.

12.14.4 Night Bell

The ShoreTel system can provide an overhead night bell on a per site basis using the audio output port associated with ShoreGear switches that provide an audio output port.

12.14.5 Intercom

A user can initiate an intercom call through a programmable button on an IP phone that has been programmed with the Intercom feature, via the ShoreTel Communicator, or via the phone by entering "*15" + extension number. Users must be configured to use the intercom feature through ShoreWare Director.

All intercom calls defeat the user's call coverage (Call Handling Mode settings) and cannot be forwarded.

An intercom call to an idle IP phone is auto-answered and connected through the called party's speakerphone. Immediately after the call is auto-answered, the called party hears an announcement tone and the calling party hears a beep tone. If the called phone was taken off-hook automatically, the switch puts the phone back on-hook when the intercom call terminates.

An intercom call to an analog phone or SoftPhone that is off-hook with no active call (for example, in hands-free mode) is auto-answered through the audio device that is currently active. If the called party is on-hook or is on an active call, the call is offered as an ordinary call, except that call coverage is still defeated.

An intercept tone (fast-busy) is played if the calling user does not have the appropriate permissions. If the called party does not accept intercom calls due to CoS permissions, the call is offered as an ordinary call.

12.14.5.1 Intercom Telephony COS Configuration

Each telephony class-of-service permissions has two additional check box settings in ShoreWare Director to configure intercom permissions.

Allow initiation for Directed Intercom/Paging—If enabled, users with this COS may make intercom calls to other users of the system. If disabled, then intercom calls cannot be made.

Accept Directed Intercom/Paging—If enabled, users with this COS may accept intercom calls. If disabled, then intercom calls are received as normal calls.

12.14.6 Call Recording

The ShoreTel system provides the capability for users to record calls. In order to use call recording, the feature must be configured in ShoreWare Director by a system administrator. Please refer to the ***ShoreTel Administration Guide*** for details on configuring this feature.

Users can use ShoreTel Communicator -Personal Access to request that a call be recorded to voice mail. Supervisors may use Agent Monitor to record an agent's call. Ordinarily, both ShoreTel Communicator and Agent Monitor will indicate when a call is being recorded, (although this behavior can be overridden with the “Silent Recording” feature to prevent agents from knowing that their calls are being recorded.)

With Silent Recording, if the call is recorded by the workgroup supervisor, the indicator does not appear in Agent Monitor. (The person invoking the recording sees the indicator—other parties do not.) In this way, calls can be silently recorded to allow operators and supervisors to hide the fact that they are recording agents' calls. This hidden behavior may be desirable when a supervisor is monitoring the telephone manners of a new employee. When the recording is silent or hidden, ShoreTel Communicator offers no visual or audible indication that the call is being recorded, and the periodic beeping sound (used to notify call participants that their calls are being recorded) is suppressed.

ShoreGear switches can support as many simultaneous recordings as there are trunk ports.

The following limitations apply to call recording:

- Call recording is only available via ShoreTel Communicator - Personal Access or a programmable button on IP phones
- Only calls on trunks (not extensions-to-extensions) may be recorded
- 2-way and 3-way calls may be recorded as long as one of the legs of the call is a trunk
- Calls to a ShoreGear Converged Conference Bridge cannot be recorded
- Recording stops when the call is parked, unparked, or transferred

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 configuring the call recording feature, you may wish to consult with legal counsel regarding your intended use.

12.14.7 Make Me Conferencing

The ShoreTel system allows up to six callers to participate in a conference call. To use the make me conference feature, you need one of the following IP phones: ShorePhone-IP110/115/212k/230/530/560/560g, and the proper Class of Service must be configured in ShoreWare Director. If you do not have an IP phone, the feature can also be used from the soft button “join” on an analog phone, in conjunction with ShoreTel Communicator. The conference ports must also be reserved on the ShoreGear switch.

The Make Me conference feature does not require a ShoreTel Conference Bridge.

12.15 ShoreGear Converged Conference Bridge

Before you connect and boot the conference bridge, you must allocate 12, 24, 36, 48, or 96 IP ports on ShoreGear voice switches using ShoreWare Director. For more information, see the ***ShoreTel Administration Guide***.

Next, determine the IP addresses that will be assigned to the conference bridge, and note the identified IP address assignments in your installation plan.

The bridge must have one IP address statically assigned for each port supported by the bridge. This requires you to identify 12, 24, 36, 48, or 96 IP addresses in blocks of 12 consecutive address according to the licensed capacity of your conference bridge.

Additionally, the bridge itself must be assigned a single static address for management and configuration access.

12.15.1 Dialing the Conference Bridge

To provide an extension for users to “dial into” their conference calls, the conference bridge requires a single number (extension) in your dialing plan. This extension is assigned to the first port of the bridge. Internal users reach the conference bridge and their conference calls by directly dialing the extension assigned to the first port. The extension is configured to distribute calls to available ports, which eliminates the need for users to dial directly into a specific port or phone number.

External callers are provided access to the bridge by configuring the appropriate trunks to be directed to the bridge. You can configure one or more of the following options:

Callers can reach the bridge through a trunk that directs all calls to the conference bridge extension. In this case, the number that external users call is the trunk’s telephone number.

The conference bridge extension can be associated with a number in your system’s DID or DNIS range to provide direct dialing to the conference bridge. In this case, the number that users call is the DID number assigned to the conference bridge.

Callers can reach the bridge by selecting the appropriate option from the system auto-attendant. In this case, the access number for the bridge is the number of the system auto-attendant.

The conference bridge is configured with up to three telephone numbers for external access. For more information, see the *ShoreTel Converged Conferencing Solution Conference Director Installation and Administrative Guide*.

12.16 ShoreTel Contact Center Solution

If you purchased a ShoreTel Contact Center Solution, you must configure an appropriate number of route points with adequate call stacks. Route points are a licensed feature. Ensure that you have sufficient licenses to support your planned deployment.

For information on route points, see the ***ShoreTel Administration Guide***. For information on the ShoreTel Contact Center Solution, please review the ShoreTel Contact Center Solution Installation Guide and the ShoreTel Contact Center Solution Administration Guide.

CHAPTER 13

Desktop Requirements

This chapter describes the hardware and software requirements for installing the end-user desktop client software.

13.1 Checklist

Review the following hardware and software requirements before proceeding to the next chapter:

Task	Description
Recommendations	page 183
Hardware Requirements	page 184
Software Requirements	page 184
Network Requirements	page 189

Table 13-1 Desktop Requirements Checklist

The installation procedures are covered in Chapter 18, starting on page 239.

13.2 Recommendations

The following recommendations will assist you in planning and installing your desktop computers for the ShoreTel Communicator applications.

Verify that each computer meets the minimum hardware and software requirements.

Install the Client for the Microsoft Networking component.

Close all applications before installing software.

Users running Windows XP Professional or Microsoft Windows Vista must have local administrative privileges to install the software.

Microsoft Outlook must be configured in Corporate or Workgroup mode for Outlook Integration to function properly. Internet Only mode is not supported.

Users should be informed of which ShoreTel Communicator application they will be using.

During fresh install or upgrade to the ShoreTel client, VSTO pre-requisites need to be installed first. The VSTO pre-requisites will be installed automatically during the ShoreTel client installation.

13.3 Hardware Requirements

Computer Hardware requirements for running ShoreTel Communicator depend on the ShoreTel Communicator version and the video call resolution. Table 13-2 displays the recommended configuration for computers running ShoreWare ShoreTel Communicator.

ShoreTel Communicator Version	Processor XP and Vista	Processor Windows 7	Max Presence Load
Communicator with Personal Access	Pentium 3 – 800 MHz	Pentium 4 - 1.6 GHz	No Presence supported
Communicator with Professional Access	Pentium 4 - 3.0 GHz with HT or Dual Core 1.6 GHz	Pentium 4 - 1.6 GHz	1 Event/Second
Communicator with Agent, Supervisor, Operator Access (<40 extension presences)	Pentium 4 - 3.0 GHz with HT or Dual Core 1.6 GHz	Pentium 4 - 3.0 GHz with HT or Dual Core 1.6 GHz	1 Event/Second
Communicator with Agent, Supervisor, Operator Access (<500 extension presences)	Dual Core 1.6 GHz	Dual Core 1.6 GHz	1 Event/Second
All Versions, VGA Video	Dual-Core 1.6 GHz	Dual-Core 1.6 GHz	1 Event/Second
All Versions, XGA Video	Core 2 Quad 2.4 GHz	Core 2 Quad 2.4 GHz	1 Event/Second

Table 13-2 Client Device Hardware Requirements

Communicator Version	Disc[a]				RAM[b]			Available RAM		
	XP	Vista	Vista	W7	XP	Vista	W7	XP	Vista	W7
Communicator with Personal Access	1 GB	1 GB	1 GB	1 GB	1 GB	2 GB	2 GB	100 MB	100 MB	150 MB
Communicator with Professional Access	1 GB	1 GB	1 GB	1 GB	1 GB	2 GB	2 GB	150 MB	150 MB	250 MB
Communicator with Agent, Supervisor, Operator Access (<40 extension presence)	1 GB	1 GB	1 GB	1 GB	1 GB	2 GB	2 GB	150 MB	150 MB	250 MB
Communicator with Agent, Supervisor, Operator Access (<500 extension presence)	1 GB	1 GB	1 GB	1 GB	1 GB	2 GB	2 GB	150 MB	150 MB	250 MB
All Versions, VGA Video	1 GB	1 GB	1 GB	1 GB	1 GB	2 GB	2 GB	150 MB	150 MB	250 MB
*All Versions, XGA Video	1 GB	2 GB	2 GB	2 GB	1 GB	2 GB	2 GB	150 MB	150 MB	250 MB

a.Disc space requirement is for installation on a system without .NET Framework installed previously. Once installed, Communicator requires less than 100 MB disc space.

b.Lists ShoreTel Communicator memory requirements during normal operation. When running other office applications on the PC in addition to Communicator, memory recommendations are 512 MB (XP) or 1 GB (Vista).

13.4 Software Requirements

13.4.1 Operating Systems

ShoreTel Communicator applications is supported on the following operating systems:

Windows XP Professional, 32 bit, SP3

Windows Vista Business, 32-bit, SP2

Windows Vista Enterprise, 32-bit, SP2

Windows Vista Enterprise, 64-bit, SP2

Windows 7 Business, 32-bit

Windows 7 Enterprise, 32-bit

Windows 7 Business, 64-bit

Windows 7 Enterprise, 64-bit

Unicode support (Double Byte Character)

Windows Terminal Server

Windows 2008 Terminal Server, 32-bit, SP2

Windows 2008 Terminal Server , 64-bit

WTS Version	Max Number of Users supported per WTS	Max Presence/Call Load [1]	RAM Memory required per client within the WTS	Minimum Processor [2]
W2008 32b SP2 Enterprise	50	2400 BHCC	100 MB	Single DualCore E5410 @ 2.33 GHZ / 16 GB RAM
W2008 64b SP2 (Non R2) Standard - Enterprise	100	5000 BHCC	200 MB	Single QuadCore E5410 @ 2.33 GHZ / 32 GB RAM

CitrixXenApp 5.0 Windows 2008, 32-bit SP2 (Isolation mode is not supported)

For ShoreTel desktop applications to function correctly, you must install the Client for Microsoft Networking.

Citrix and Windows Terminal Server (WTS) technologies enable processing for multiple users to be aggregated on a single Windows computer. The single Windows computer is a process and disk sharing server for multiple users that have lightweight or thin graphics stations on their desktop. Citrix communicates between the server and clients using the ICA protocol, whereas Windows Terminal Server uses the RDP protocol.

For configuration and support information on c WTS servers running ShoreTel Communicator clients, see Appendix E, starting on page 277.

13.4.2 Internet Browsers

ShoreWare Web Access call handling application requires one of the following browsers:

MS Internet Explorer 7.0

MS Internet Explorer 8.0

Web Client

Safari 4.0 on Macintosh
MS Internet Explorer 8.0
Firefox 3.6 on Windows
Adobe Flash 9,10

13.4.3 Microsoft Outlook Integration

The following are required when the Microsoft Outlook Integration feature is used:

Outlook 2003, SP2

Outlook 2007, SP2

Additional Requirements for Microsoft Outlook Integration include:

Microsoft Outlook must already be installed as the user's email before installing Outlook integration features (see the installation procedure in Chapter 18, starting on page 239).

Outlook must be configured for mode (supporting multiple mail service providers) and not for mode before installing Outlook integration features.

Automatic Call Handling with the Microsoft Outlook Calendar requires an optional component of Microsoft Office called Collaborative Data Objects.

The Collaboration Data Object must be installed in order for Microsoft Outlook calendar integration to work.

13.4.3.1 Offline Call Handling

ShoreTel 11 can integrate with Microsoft Outlook 2007 to allow Offline Call Handling Modes (CHM). Once configured, a user's CHM will automatically change, based on the user's Outlook calendar, even when Outlook is not currently opened. The Call Application Server (CAS) is responsible for handling the call handling mode changes and user configurations associated with this feature.
Microsoft Outlook 2007 Plug-in with Offline Call Handling Modes requires the installation of the following components on the client system:

NET Framework 3.5

Outlook 2007 Primary Interop Assemblies

Visual Studio Tools for Office Runtime 3.

Outlook CAS Interface add-in

13.4.3.2 ShoreTel Communicator Integration with Outlook Implementation

Outlook integration may be installed from ShoreTel Communicator when the user navigates to the Options and Preferences Dialog menu and selects the Outlook option. Users need to quit Outlook to install the plug-in.

Note: Once the plug-in is installed, it is not necessary to restart the computer.

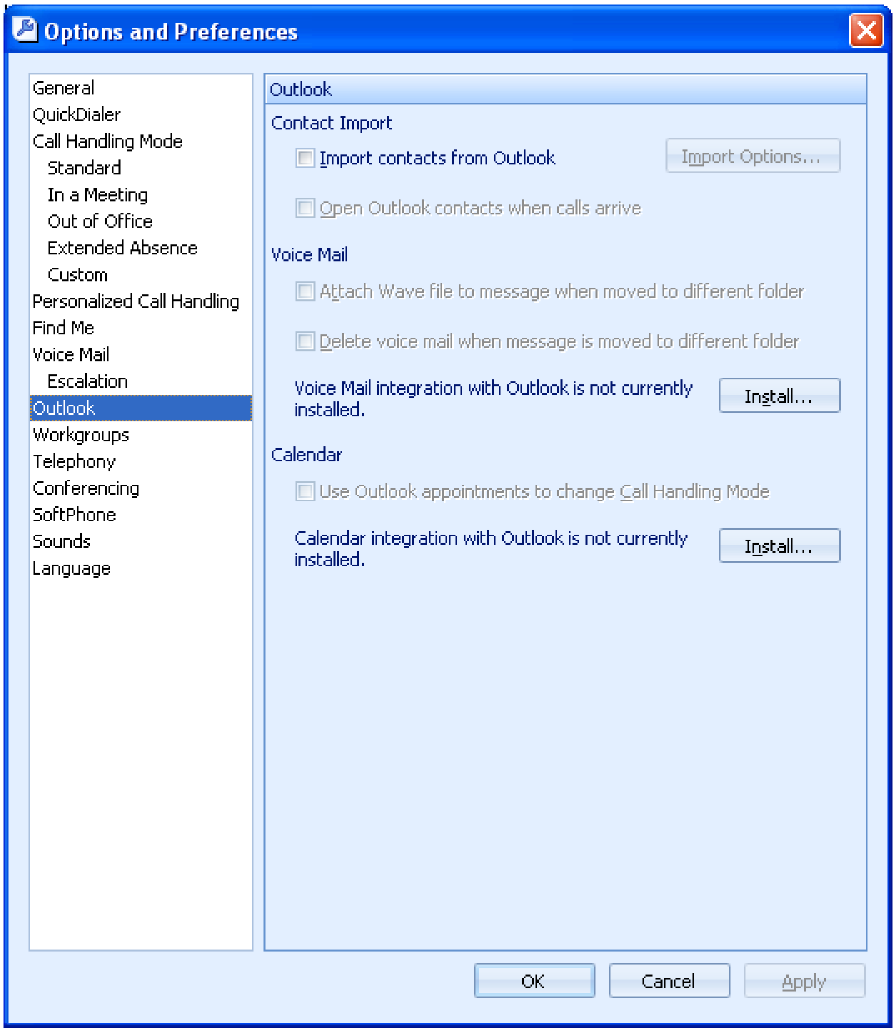


Figure 13-1 ShoreTel Communicator Options - Outlook Install

13.4.3.3 Outlook Call Handling Modes

ShoreTel now supports a simple drop down list located in the ShoreTel Ribbon Group of the Appointment Ribbon. A new ShoreTel Ribbon Group is added to the Appointment Ribbon. This ribbon group will contain the call Call Handling Mode menu.

Figure 13-2 Microsoft Plug-in - Call Handling Modes Drop down

When the drop down arrow on the button is selected, the Call Handling Modes options will appear. Users can set their call handling mode using the drop down arrow on the button is selected.

Figure 13-3 Call Handling Mode Button

Call Handling Modes Options Dialog

The Call Handling Modes setup is located in the Outlook Options page under the ShoreTel Call Handling Schedule tab.

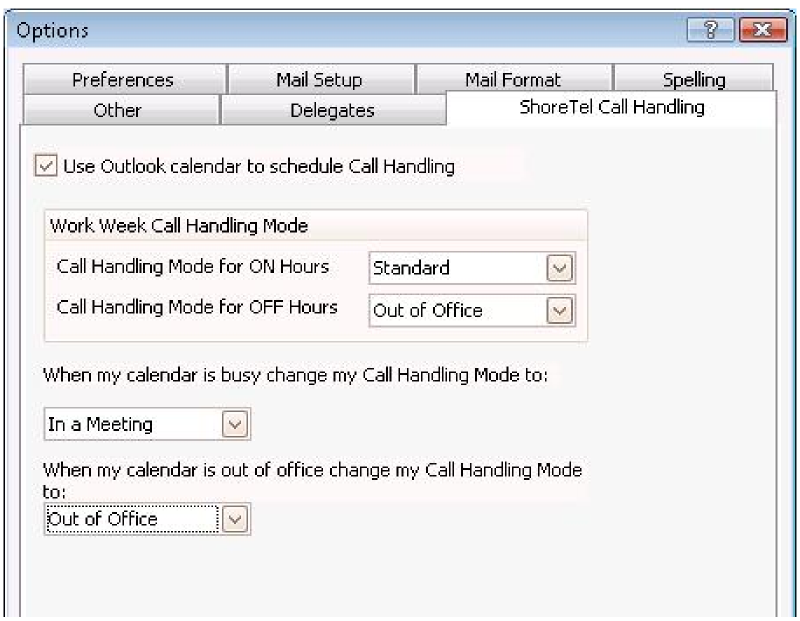


Figure 13-4 Outlook Options - ShoreTel Call Handling Schedule Tab

Calendar Busy/Out of Office Options

The Calendar Busy and Out of Office options are configurations that the user can use to set their call handling mode, based upon the appointment item's busy status configuration. If the Outlook appointment item's busy status configuration is set to show time as busy when the appointment starts, the CHM will automatically change to "In a Meeting." If the busy status configuration is set to show time as out of office, the CHM will change to “Out of Office.”

Note: These options are overridden when a user manually changes their Call Handling Mode in the appointment ribbon.

Working Offline

Users should NOT change their appointments when Outlook is offline (not connected to the network). If the user modifies their schedules while offline, the web meeting or Call Handling Mode features will not be available and will not be uploaded to Call Application Server.

Localization

The Outlook CHM add-in may be customized for language. Languages that are currently supported with ShoreTel Communicator are supported for the Outlook integration modules.

These modules will automatically pick the current Outlook interface language if an Add-in dll has been provided in that language. If no Add-in dll is provided for that language, it will fall back to the system default. For example: if a user has a German version of Outlook, the Call Handling Mode drop down, will display all text in German, provided that the Add-In has been localized for German.

13.4.4 Microsoft Updates on the Server

ShoreTel weekly updates test systems with the latest Microsoft desktop patches. When releasing a new build, ShoreTel publishes Build Notes that lists the Microsoft patches that are certified against the build. ShoreTel also highlights software changes required by the MS patches.

The conservative approach is to turn off regular MS updates until you review the detailed certification provided with each release.

13.5 Network Requirements

Personal computers running ShoreTel Communicator software must be networked to the ShoreWare server. See Chapter 9, starting on page 107, for bandwidth requirements.

13.6 Virus Protection Desktop Systems

ShoreTel allows the use of industry standard virus protection software on desktop systems running the client application.

13.7 VMware Virtual Environment for Main and DVS Servers

13.7.1 Versions

The following versions of VMware will be supported:

VMware vSphere 4 (ESX 4 / ESXi 4)

13.7.2 Configuration (For Main Server Only)

High Availability

VMotion

Note: Only server running on VMware for the core platform is supported. The ECC, CSTA, and other TPP and professional services applications are not supported on the ShoreWare virtual servers.

CHAPTER 14

Site Requirements and Preparation

This chapter provides information about preparing your site for the ShoreTel system, including concerns such as physical space, environment, and cabling.

14.1 Checklist

Review the following site requirement topics before proceeding to the next chapter:

Task	Description
Recommendations	page 191
Voice Switch Requirements	page 192
Racks and Cabling	page 197
A 19-inch data rack, shelf, and modular patch panels can be purchased from most major electrical suppliers.	page 198

Table 14-1 Site Requirements and Preparation Checklist

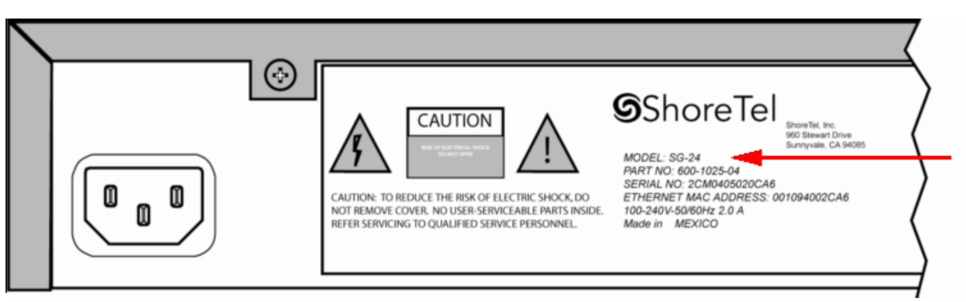
14.2 Recommendations

The following recommendations will assist you in planning and preparing your site for the ShoreTel system.

Hire a cabling contractor to install your racks, patch panels, and cabling.
Have an RJ-48C cable ready for each voice switch.

14.2.1 Switch Models

You can locate the model number of your switches on the rear panel as shown in Figure . This document distinguishes between switches based on the model number and the number of RU's the switch occupies.



ShoreGear 120 Model Number Label

Refer to Appendix H for a information on all ShoreGear switches, including capabilities, connectors, and LED behavior.

14.3 Voice Switch Requirements

This section includes requirements for mounting the ShoreGear voice switches, along with other switch-related requirements and specifications.

14.3.1 Physical Requirements

The ShoreGear voice switches are designed to be mounted in a standard 19-inch rack. Table 14-2 shows the specifications for each voice switch. For more information refer to the , included with each ShoreGear voice switch.

Parameter	ShoreGear 120 ShoreGear 60 ShoreGear 40	ShoreGear T1 ShoreGear E1
Dimensions (H x W x D)	1.72” x 17.16” x 14.28 “ 43.68 x 435.86 x 362.71 mm	1.72” x 17.16” x 14.28 “ 43.68 x 435.86 x 362.71 mm
Rack mount units	1 RU	1 RU
Mounting position	Front, Center	Front, Center
Weight	9 lbs 4.08 kg	8 lbs 3.62 kg
Maximum stacked per shelf	3 switches	3 switches

Table 14-2 ShoreGear Voice Switch Physical Specifications

Table 14-3 shows the latest hardware line, designed to increase port density.

Parameter	Dimension
Dimensions (H x W x D)	1.69” x 8.39” x 14.28 “ 43 x 213 x 378 mm
Rack mount units	1 RU
Mounting position	Front, Center
Weight	5.3 lbs 2.4 kg
Maximum stacked per shelf	6 switches

Table 14-3 Half-Width ShoreGear Voice Switch Physical Specifications

14.3.2 Input Power

For backup purposes, ShoreTel recommends that all ShoreGear voice switches and the ShoreWare server be connected to an uninterruptable power supply (UPS). This ensures that telephone service will continue in the event of a power interruption.

Table 14-4 shows the power requirements for the full-width ShoreGear voice switches.

Table 14-5 shows the power requirements for the half-width ShoreGear voice switches.

Parameter	ShoreGear 120 ShoreGear 60	ShoreGear 40	ShoreGear T1 ShoreGear E1
Input voltage	100–240 VAC 50–60 Hz	100–240 VAC 50–60 Hz	100–240 VAC 50–60 Hz
Current consumption @110 VAC (maximum)	2A max	1A max	1A max
Number of grounded 110 VAC outlets per switch	1	1	1
Power consumption (typical)	90W typ	50W typ	50W typ

Table 14-4 ShoreGear Voice Switch Power Input (Full-Width Switches)

Parameter	ShoreGear 90 ShoreGear 90BRI ShoreGear 50 ShoreGear 30	ShoreGear 220T1 ShoreGear 220T1A ShoreGear 220E1 ShoreGear T1k
Input voltage	100–240 VAC 50–60 Hz	100–240 VAC 50–60 Hz
Current consumption @110 VAC (maximum)	1A max	1A max
Number of grounded 110 VAC outlets per switch	1	1
Power consumption (typical)	40W	17W

Table 14-5 ShoreGear Voice Switch Power Input (Half-Width Switches)

14.3.3 Power and Heat Dissipation

The voice switches dissipate power and heat. ShoreTel recommends that you use the information provided in Table 14-6 and Table 14-7 to help calculate the ventilation requirements of the equipment room.

Parameter	SG 120/24	SG 60/12	SG 40/8	SG T1	SG E1
Power dissipation (typical)	90 W typ	90 W typ	50 W typ	50 W typ	50 W typ
Heat dissipation	215 BTU/hour	140 BTU/hour	85 BTU/hour	61 BTU/hour	65 BTU/hour

Table 14-6 ShoreGear Voice Switch Power and Heat Dissipation (Full-Width Switches)

Parameter	SG 90 SG 90BRI SG 50 SG 30	SG 220T1 SG 220T1A	ShoreGear 220E1
Power dissipation (typical)	40 W typ	40 W typ	17 W typ
Heat dissipation	137 BTU/hour	137 BTU/hour	58 BTU/hour

Table 14-7 ShoreGear Voice Switch Power and Heat Dissipation (Half-Width Switches)

14.3.4 Environmental Requirements

The ShoreGear voice switches require that the environmental specifications provided in Table 14-8 be met.

Parameter	Specification
Operating temperature	0° to 50° C (32° to 122° F)
Operating humidity (non-condensing)	10% to 90%
Storage temperature	–30° C to 70° C (-34.4° to 158° F)

Table 14-8 ShoreGear Environmental Specifications

14.3.5 Reliability and Availability

Each ShoreGear voice switch is an embedded product with no moving parts other than a highly reliable fan. In addition, the power supply contained within the voice switch has a very high individual mean time before failure (MTBF), as shown in Table 14-9.

Voice Switch	Predicted MTBF (hours)	Demonstrated MTBF (hours)	MTTR (hours)	Availability
ShoreGear 120	84,570	320,142	1	99.9997%
ShoreGear 60	90,956	152,388	1	99.9993%
ShoreGear 40	132,302	314,557	1	99.9997%
ShoreGear T1	158,229	312,709	1	99.9997%
ShoreGear E1	154,229	312,709	1	99.9997%
MTBF = Mean time before failure MTTR - Mean time to repair Availability = %uptime/time = MTBF/(MTBF+MTTR)				

Table 14-9 ShoreGear Voice Switch Dependability

Since the ShoreTel system is plug-and-play, a switch can be replaced in minutes.

Distributed call control software means there is no system-wide single point of failure. If a single ShoreGear voice switch fails, all the other voice switches continue to operate.

Table 14-10 shows the reliability information for the ShorePhone phones. Hourly numbers shown are based on demonstrated reliability (as opposed to calculated).

Phone	MTBF hours (calculated)	MTBF hours (demonstrated)
IP110	64,800	490,000
IP115	N/A	TBD
IP210	62,100	240,000
IP212k	58,200	350,000
IP230	58,200	350,000
IP265	N/A	TBD
IP530/560	56,300	360,000
IP560g	56,400	TBD
IP565g	N/A	TBD
BB24	72,600	TBD

Table 14-10 ShorePhone IP Phone Dependability

14.3.6 Memory and Processing

Each ShoreGear voice switch has the same memory and processing (see table below).

Type	Details
Flash Memory	16 MB
Random Access Memory	128 MB
Main Processor	PowerPC 8245
Digital Signal Processor	Texas Instruments 5409A

Table 14-11 ShoreGear Voice Switch Memory and Processing

14.3.7 Connectors

Table 14-12 summarizes all of the connectors on the ShoreGear voice switches. Diagrams showing where these connectors are located are provided later in this chapter.

Port/Connector	SG 120/24 SG 60/12 SG 40/8	SG T1 SG E1
Power	110 VAC	110 VAC
Ethernet	2 RJ-45	2 RJ-45
Analog telephone/trunk	RJ-21X male 0–2,000 feet[a]	— —
Digital trunk	—	RJ-48C
T1 trunk monitor	—	RJ-48C
Audio input (Music on Hold)	3.5 mini-mono	—
Audio output (Paging, Night Bell)	3.5 mini-mono	—
Maintenance	DB-9 female	DB-9 female

Table 14-12 ShoreGear Voice Switch Connectors

a. 2000 ft. length uses 26AWG wire.

14.3.7.1 Power Cabling

Each ShoreGear voice switch comes equipped with a standard 110 VAC modular power cord. A localized modular power cord can be ordered from ShoreTel. ShoreTel recommends that every ShoreGear voice switch, as well as the ShoreWare server, be connected to an uninterruptable power supply (UPS).

14.3.7.2 Ethernet Cabling

Each ShoreGear voice switch has two RJ-45 connectors that provide an auto-sensing 10/100M Ethernet interface. These are connected to the local area network using standard Category 5 cabling.

ShoreGear voice switches come with two network interfaces, LAN1 and LAN2, allowing for a network fault tolerant deployment. You can connect to either or both connectors; there is no primary/secondary relationship. When both are connected, only one will be active at any time. If the currently active interface loses the link, the alternate interface becomes active. Both interfaces will use the same MAC Ethernet address, and IP address.

There are two levels of fault tolerance. To protect against Ethernet switch failure, connect LAN1 and LAN 2 to separate Ethernet switches. To protect against port or cable failure, connect LAN1 and LAN2 to separate ports on the same Ethernet switch.

10 Base-T and 100 Base-T can typically support up to 100 meters.

14.3.7.3 IP Phone Cabling

Each ShorePhone IP phone has an RJ-45 connector that provides an auto-sensing 10/100M Ethernet interface. This is connected to the local area network using standard Category 5 cabling.

10 Base-T and 100 Base-T can typically support up to 100 meters.

14.3.7.4 Analog Telephone and Trunk Cabling

ShoreGear voice switches that support analog protocols provide an RJ-21X male connector for mass termination of the telephones and trunks. This should be connected using a standard 25-pair cable. ShoreTel recommends using the RJ-21X and connecting to a patch panel to provide simple moves, adds, and changes.

Telephones can be supported from 0 to 2,000 feet from the voice switch over standard cabling. Use larger gauge wires for longer distances.

It is recommended that an analog telephone be provisioned in the equipment room for troubleshooting purposes.

Pinouts of the ShoreGear switches are shown in the section Appendix G, starting on page 303.

14.3.7.5 Digital Trunk and Trunk Monitor Cabling

ShoreGear voice switches that support digital trunks have an RJ-48C connector as the telco interface to the T1/E1 trunk from the telephone service provider.

These voice switches provide an internal Channel Service Unit (CSU).

ShoreGear voice switches that support T1 and E1 trunks have an additional RJ-48C connector that is wired to the telco interface for the purpose of troubleshooting the T1 or E1 interface with specialized test equipment. This connector is normally not used.

14.3.7.6 Audio Input (Music on Hold) Cabling

Various ShoreGear voice switches have a 3.5 mm mini-stereo input connector that provide music or some other recording to callers when they are on hold. The input port supports low-level line audio from a preamplifier or mini-CD player, at 47 kΩ nominal impedance. The audio input cable can be up to 10 feet long. Refer to Appendix G, starting on page 303, to determine the voice switches that provide the 3.5 mm mini-stereo input connector.

The audio input port on the ShoreGear voice switches is a mono connection. If you connect a stereo input, the stereo signal is converted to a mono signal.

To minimize bandwidth, music on hold is not streamed across the wide area network, so you will need one music source per site.

The music and music source are not included with the ShoreTel system.

WARNING In accordance with United States copyright laws, a license may be required from the American Society of Composers, Authors, and Publishers, or a similar organization, if radio or TV broadcasts are played for music on hold. As an alternative, an ASCAP-approved CD or tape can be used. ShoreTel, Inc. disclaims any liability out of failure to obtain such a license.

14.3.7.7 Audio Output (Paging and Night Bell) Cabling

Various ShoreGear voice switches have a 3.5 mm mini-stereo audio output connector for overhead paging and night bell on a per site basis. The audio output port provides low-level line audio with a sufficient input level for a typical amplifier. The paging port output is about one volt peak to peak, similar to the line output of a CD player, and can drive inputs that are 600 ohms or higher. Refer to Appendix G, starting on page 303, to determine the voice switches that provide the 3.5 mm mini-stereo input connector.

The audio output is mono signal. If you use a stereo jack, the signal is available on one channel, but the other channel will be silent.

This is a single-zone paging system. If more zones are required, see the application note on ShoreTel's online knowledge base.

14.3.7.8 Maintenance Cabling

ShoreGear voice switches support a maintenance port for connection terminal using a standard DB-9 female connector. This maintenance port is typically used only when assigning networking parameters if DHCP or BOOTP is not being used.

14.4 Racks and Cabling

14.4.1 General Cabling Overview

The diagram in Figure 14-1 highlights the key components with respect to cabling for your voice network.

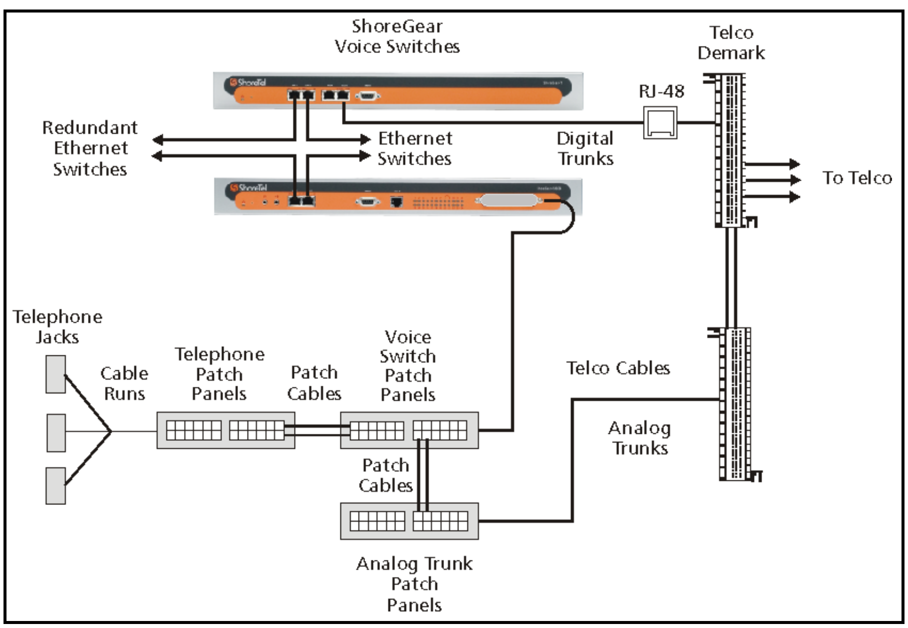


Figure 14-1 Cabling Overview

Starting from the lower left in this diagram, the telephone cabling is organized as follows:

A telephone jack (RJ-11) is provided for each telephone.

Telephone cabling (Category 3 or better) is terminated on the telephone jack and runs back to the equipment room to a modular connector (RJ-21X) on a telephone patch panel.

The telephone patch panel provides a flexible cable management solution for the telephone cabling. The patch panel has RJ-21X connections for the telephone cabling and RJ-11 connections on the front.

Patch cords are connected from the telephone patch panel (RJ-11) to the voice switch patch panel (RJ-11).

The voice switch patch panel provides a flexible cable management solution for the voice switches. The patch panel has RJ-21X connections running to the voice switches and RJ-11 connections on the front.

Starting from the right in Figure 14-1, the trunk cabling is organized as follows:

The digital (T1/E1) and analog trunks are terminated on a punch-down block.

The digital service is further terminated at a service provider demark with an RJ-48 connector.

An RJ-48 cable from the T1/E1 demark connects to the ShoreGear T1 or ShoreGear E1.

The analog service is cross-connected to a modular (RJ-21X) punch-down block.

A telco cable is connected to the modular (RJ-21X) punch-down jackand runs to a modular connector (RJ-21X) on an analog trunk patch panel.

Like the telephone cabling, patch cords are connected from the analog trunk patch panel (RJ-11) to the voice switch patch panel (RJ-11).

As an alternative, patch panels can be replaced with punch-down blocks. This may be more cost-effective but is less flexible.

14.4.2 Rack Overview

Figure 14-2 shows a typical rack installation.

A 19-inch data rack, shelf, and modular patch panels can be purchased from most major electrical suppliers.

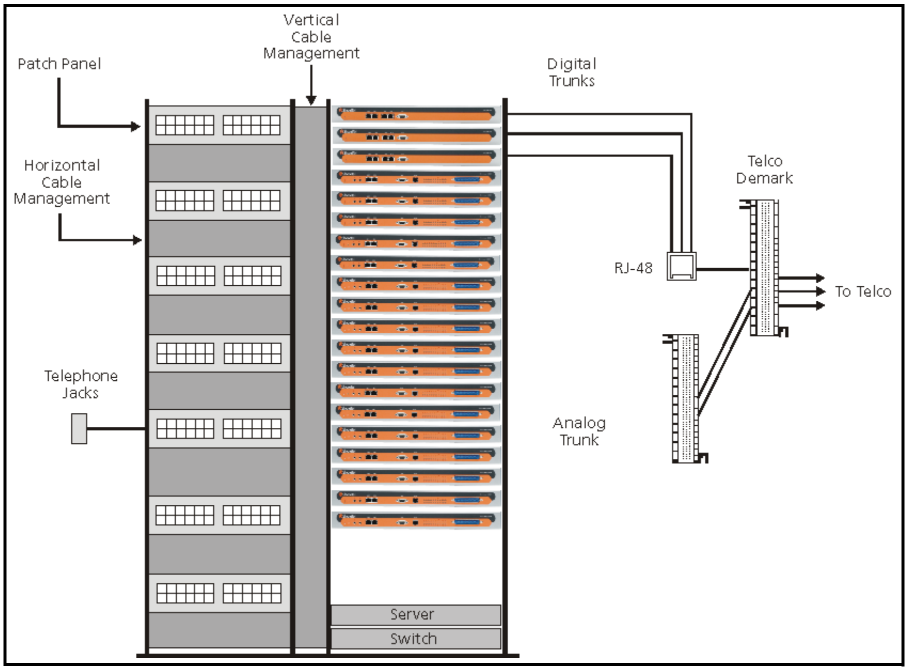


Figure 14-2 Rack Installation

CHAPTER 15

Legacy Integration

ShoreTel provides a migration solution from a legacy TDM-based voice network into the voice-over-IP ShoreTel system. You can handle line growth and enable a migration of users from the legacy system to the ShoreTel IP PBX by deploying the ShoreTel system at one location in a multi-location enterprise, or side-by-side with a legacy PBX at a single location.

Integrating the ShoreTel system with your legacy PBX's allows users on the different systems to communicate with each other effectively for both phone calls and using voice mail.

With an integrated voice network, you can:

- Simplify communications for your users with an enterprise-wide coordinated dialing plan using extension dialing.
- Exchange voice mail messages between users on different sites using different voice mail systems. Standard commands such as compose, forward, and replay extend the value of your different voice mail systems.
- Consolidate trunks with different traffic types to leverage different service provider rates.
- Reduce service costs by redirecting inter-site calls across your IP network.

15.1 Checklist

Review the following topics before proceeding to the next chapter:

Task	Description
Coordinated Dialing	page 202
Trunk Requirements	page 203
Coordinated Dialing Plan	page 203
PSTN Services	page 204
Multi-Site Integration	page 204
Single Site Integration	page 204
Consolidated Long Distance	page 205
Voice Mail Integration	page 205
System Requirements	page 221

Table 15-1 Legacy Integration Checklist

Task	Description
Connection Cable	page 221
Administration and Configuration	page 222
Trunk Configuration	page 222

Table 15-1 Legacy Integration Checklist

15.2 Introduction

A digital trunk "tie" line integrates the ShoreTel system with a legacy PBX. The connection is between the legacy system's PRI interface and the PRI interface of a ShoreGear switch located anywhere in your IP network.

There are four different types of activities that occur on the interface.

Calls from ShoreTel users or applications to an extension located on the other system are routed across the tie trunk. When a call is placed, the trunk is accessed and the ShoreTel system sends the configured number of digits to the PBX identifying the called extension.

Calls from users on the legacy system or from trunks, or other applications on the legacy PBX, are routed across this interface. When the legacy user places their call, the legacy system accesses the trunk and then sends the digits as DNIS.

Outbound calls from users or applications on the ShoreTel system can be routed across the trunk to the legacy PBX. When a call is placed, the trunk access code or trunk configuration of the connection to the legacy PBX indicates the outbound call is to be placed to the PBX.

Calls between the ShoreTel and legacy system's voice mail applications are carried across the trunk connecting the two systems. The voice mail systems make calls to configured destinations on the other system to send voice mail messages to users on the other system.

A tie trunk is not required to enable voice mail or AMIS integration. The two voice mail systems can communicate by dialing each other via the PSTN. In general, when a tie trunk is in place, AMIS calls should be routed via the trunk to reduce PSTN costs.

The connection between the two systems can be provided by either T1 trunks or by a PRI interface. ShoreTel recommends that you use PRI to enable calling number information exchanges between the two systems.

15.3 Coordinated Dialing

Coordinated dialing allows users to dial between the systems using extension-to-extension dialing as well as enabling consolidation of inbound and outbound services. To effectively plan the integration, consider the following items:

Expected call traffic between the two systems to provide sufficient trunking
Current numbers of extensions and extension lengths at both systems
Service plans to determine which PSTN services are provided at each voice system
The type of legacy PBX equipment integrated with the ShoreTel system

15.4 Trunk Requirements

The number of digital trunks required between the ShoreTel system and the legacy PBX depends on the expected traffic between the two systems. To determine the number of trunks, you need to estimate the number of calls per hour that are placed between the two systems. When estimating the call volume between the two systems, consider the following:

- The volume of direct calls between users on the two systems
- Traffic related to Automated Call Distributor (ADC) calls
- Outbound call volume (i.e. when outbound trunking to the PSTN is provided by one of the systems for all users, such as a PSTN trunk connected to the legacy PBX that provides long distance services for users on both the legacy and ShoreTel system)
- Inbound call volume (i.e. when inbound services are provided by one system to all users)

Additionally, you can rely on the estimated calls-per-hour number to determine the number of trunks to configure between the two systems.

For more information on trunk requirements, see Chapter 5, starting on page 67.

15.5 Coordinated Dialing Plan

With legacy integration, users on both systems can dial one another using abbreviated or extension dialing. This includes dialing from applications on the systems, such as the ShoreTel voice mail application, and would also include forwarding a call to an assistant at an extension on the legacy PBX. To determine the coordinated dialing plan configuration, you must identify the current numbering of users on both systems. For example:

- When the systems are located together, extensions can normally be assigned from a single numbering plan, or from a single DID number range provided by the local carrier. In this case, the extensions on the two systems are assigned such that there is no overlap using the desired extension length.
- When systems are at different locations, each system's numbering plan is often based on the DID range supplied by the local telephone company. In this case, overlap of the extension ranges can occur at the currently used extension length.

For example, consider the following situation.

- One location is assigned DID range 408-555-2000 through 2999
- The second location is assigned range 650-333-2500 through 2799
- The systems currently use four-digit dialing matching the trailing 4 digits of the DID numbers.

In this case, there are users on both systems currently assigned extension 2500. To provide a coordinated dialing plan across the systems, the extensions must be adjusted to make them unique system-wide. In the integration, four-digit extensions that overlap are made unique by increasing the extension length across the system. When the extension length is increased, the first digit becomes the "system" number and the remaining digits are the "extension." In the above example, the extension length would be increased to five-digit dialing, and at the first location would be extensions 52000 through 52999, while users at the second location would be assigned extensions 32500 through 32799.

The extensions on all systems that are integrated together should be configured to be the same length.

Be sure to document the planned integrated dialing plan prior to configuring the systems to streamline the configuration process. Information to take note of is provided in the following template:

	System One	System Two
Location		
DID Range		
Local Extensions (Prefix + Number)		
Remote Extensions (Prefix + Number)		

Table 15-2 Dial Plan template

15.6 PSTN Services

The number of trunks, your integration plan, and the overall system design includes the provisioning of services across the network. PSTN services can be provided at both systems in the integration or consolidated together on one system.

15.7 Multi-Site Integration

When the systems are located at different sites, both systems should have local trunking for both inbound and outbound calls. Local inbound numbers make it easy for nearby customers to reach you, while local outbound trunks allow you to save on telephone charges by using local services at the site.

In this configuration, the trunk lines connecting the systems are used for the inter-site calling between extensions or applications on the two systems. The interfaces on the two systems are configured to dial out to the remote or off-system extensions, and to accept incoming calls using DNIS.

The ShoreGear voice switch that connects to the legacy PBX should be located at the site with the legacy PBX. This leverages the IP network to extend the calls to the other sites with the ShoreTel system.

15.8 Single Site Integration

When the systems are located at the same site, it is not required that both systems be connected to the PSTN. The systems can be configured to best match your requirements.

In a single site configuration, the PSTN connections for inbound calls can be connected to each system. In this environment, the trunks connecting the two systems are configured to dial out the remote or off-system extensions and to accept incoming calls using DNIS.

Alternatively, inbound services can be consolidated on either the ShoreTel system or the legacy PBX. In this environment, calls to users on the other systems are forwarded to the remote or off-system extensions through the trunk lines connecting the systems.

When all inbound trunks are consolidated on the ShoreTel system, the trunks are configured to support off-system extensions within the range of extensions on the other PBX.

When all inbound trunks are configured on the legacy PBX, the trunks on the ShoreTel system are configured to support inbound services with call routing to the extensions on the ShoreTel system.

When DID numbers are already in place on one of the PBX's which will be connected, ShoreTel recommends that the inbound DID service not be moved or split between the systems but configured to remain on the system where they are currently configured and have calls to users on the other system forward across the connecting trunks.

In the single site configuration, ShoreTel recommends that services for outbound calls be connected to the legacy PBX. In this configuration the trunk interfaces on the s system are configured to support outbound local and long distance dialing while the interface on the PBX is configured to route the received outbound calls.

15.9 Consolidated Long Distance

Long distance calls can be consolidated into a single PSTN interface across both the ShoreTel system and the integrated legacy PBX. In this configuration, you gain the benefits of reduced long distance rates by consolidating all your enterprise's long distance calls into a single carrier. When it is required, the outbound long distance trunks are connected to the legacy PBX and the ShoreTel system is configured to route long distance calls outbound across the digital trunk connecting the systems.

15.10 Voice Mail Integration

The primary issue with voice mail integration is they are often proprietary and the interfaces defined to connect the same and disparate systems are very old, complex and difficult to implement. In fact, many voice systems from the same vendor are not connected. The interface with which most customers are familiar is AMIS. This is an analog interface that has been around for a long time, but is a real challenge to implement and can be very expensive from legacy voice mail providers. It is not uncommon to pay $10,000 per site for this capability. Another widely-used interface, Simplified Message Desk Interface (SMDI), was developed in the days when the PBX and voice mail systems were separate systems. It operates on a serial link between a PBX and voice mail system and allows them to work together. ShoreTel supports both AMIS and SMDI protocols for voice mail integration.

15.10.1 AMIS Protocol Support

The ShoreTel system sends and receives voice mail messages to and from legacy voice mail systems using AMIS protocol Version 1 - Specification February 1992. To send voice mail messages to remote AMIS sites, ShoreTel dials the access phone number for the remote system. Likewise, to receive voice messages from a remote system, the remote system must know the number to dial into the ShoreTel system. To reach the ShoreTel system, the remote system must be configured to dial any 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, users cannot send replies. To send outbound AMIS voice mail, you must define AMIS System profiles in ShoreWare Director.

ShoreTel negotiates the setup, handshaking, and teardown of AMIS system calls. Each voice mail requires a call over the trunk group defined for the AMIS delivery and call-back numbers.

To simplify AMIS systems and increase usability:

Use the same extension length across your enterprise.

Use off-system extensions to match remote users' mail boxes with their extension numbers.

Assign each system a System ID to identify the remote site location

For more information on AMIS systems, see the ***ShoreTel Administration Guide***.

15.10.2 SMDI Protocol Support

The ShoreTel product supports the SMDI protocol, enabling seamless integration of ShoreTel equipment with legacy phone systems and enabling a smooth migration toward an all-IP telephony solution.

15.10.2.1 A little history

The SMDI protocol evolved at a time when voice mail services and PBX services were provided by separate physical devices. Over the years, manufacturers have managed to offer both PBX and voice mail services within a single device, and the need for SMDI has diminished. However, the protocol can still be useful in situations where newer equipment will be integrated into a network of older devices.

15.10.2.2 How it works

SMDI enables the separate devices that provide PBX and voice mail services to share information over an out-of-band serial cable connection. The PBX shares information with the voice mail system about incoming calls. The following information is passed to the voice mail system:

who the call is from
where the call is going (i.e. user extension)
the reason the call is going to voice mail instead of being answered

In response, the voice mail system returns a notification to the PBX that a message was left on the voice mail server. The PBX system then uses this information to alert the user by turning on the "message waiting" light on his or her phone.

15.10.2.3 Configurations of integrated equipment

With SMDI support, there are essentially two possible ways the ShoreTel and legacy equipment can be configured:

– The legacy system provides voice mail services while the ShoreTel system acts as the PBX.

– The ShoreTel system provides voice mail services while the legacy system acts as the PBX.

15.10.2.4 Additional details

A group of analog trunks from the ShoreTel system is used to access the legacy voice mail system (the ShoreTel system is on the extension side of the trunks). The ShoreTel voice mail application manages the group of outgoing extensions. The ShoreTel server can provide digit translations if the legacy voice mail and ShoreTel system have different extension lengths.

Figure 15-1 shows the ShoreTel system providing PBX services and the legacy equipment providing voice mail services.

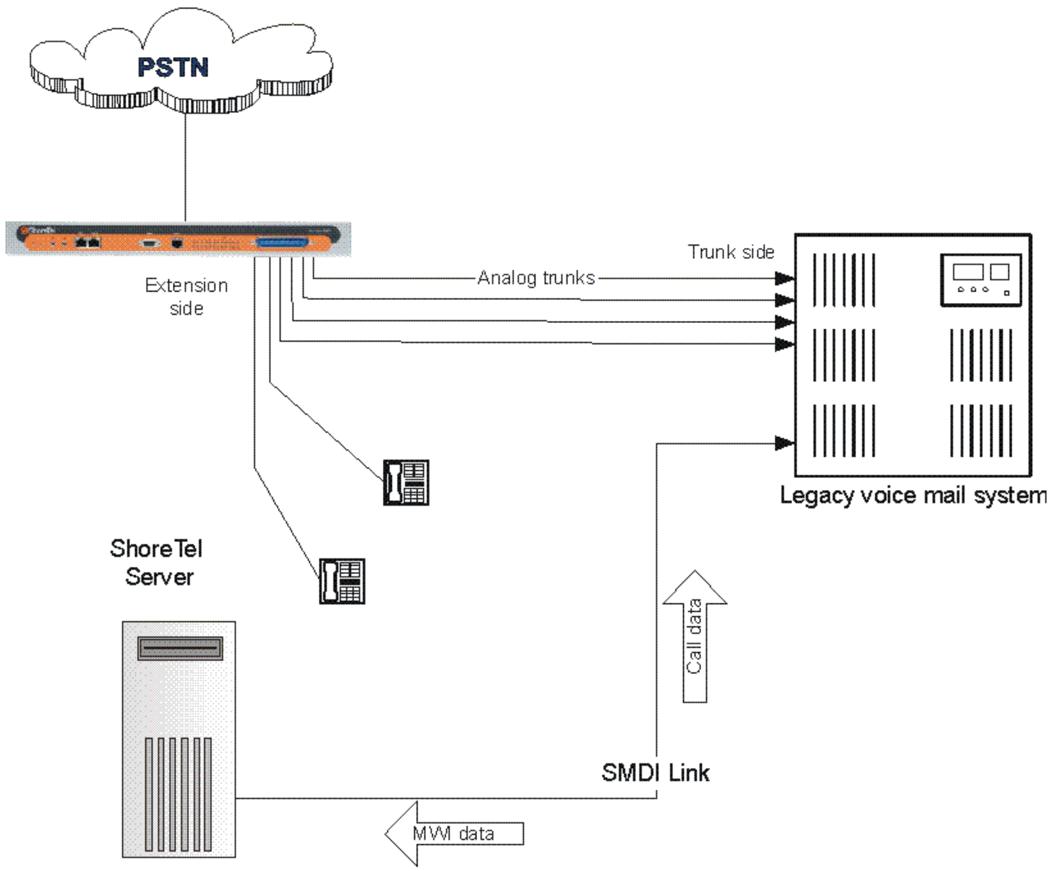


Figure 15-1 External Voice Mail with ShoreTel as PBX

Figure 15-2 below shows the legacy system providing PBX services and the ShoreTel equipment providing voice mail services.

15.10.2.5 Details

Figure 15-2 shows a ShoreTel switch connected to a legacy PBX through several analog trunks. These phone lines carry voice information from the PBX to the voice mail server. Signaling information is carried out-of-band on the separate serial line (near the bottom of the illustration).

A ShoreTel voice mail server is connected through a serial cable to a PBX link device. (The PBX link device provides the basic SMDI services that were not included in some of the older legacy PBX devices. This device must be purchased separately and configured per the manufacturer's instructions.)

The ShoreTel server and PBX link exchange information. The PBX link sends call data to the ShoreTel voice mail server, and the call data contains information related to the source and destination of the phone call, and provides information about why the call is going to voice mail (e.g. user did not answer, line was busy, etc.).

The ShoreTel server, in return, sends MWI (Message Waiting Indicator) information that is used by the legacy PBX to turn on the message-waiting mechanism on a user's phone to let her know she has received a message.

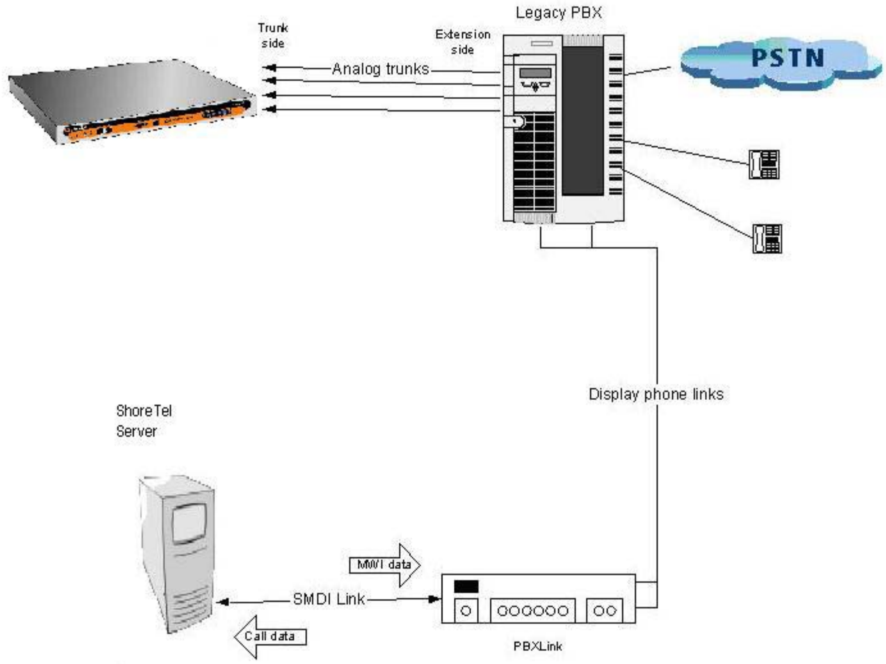


Figure 15-2 ShoreTel Voice Mail with legacy PBX

15.10.2.6 Information Transferred via SMDI

The COM port is used to send call information between the ShoreTel system and the legacy voice mail system. The SMDI protocol transmits the following call information from the ShoreTel system to the legacy system:

Message desk number: 1-999
Logical Terminal number (terminal identifier): 1-9999
Call type (All, Busy, Direct, No Answer, Unknown)
Called party
Calling party

The SMDI MWI protocol transmits the following information from the legacy voice mail system to the ShoreTel system:

Message waiting indication control
Extension
On/Off indication

15.10.3 Configuring Legacy Voice Mail Integration Using SMDI

As mentioned before, there are two modes of operation with respect to integrating a ShoreTel system and a legacy system:

External Voice Mail Configuration - In this configuration, the legacy system provides voice mail services while the ShoreTel system acts as PBX for users.

ShoreTel Voice Mail Configuration - In this configuration, the ShoreTel system provides voice mail services while the legacy system acts as a PBX for users.

The former of these two operational modes (External voice mail) is discussed below, while the procedure for the latter configuration (ShoreTel voice mail) follows in Section 15.10.4 on page 214.

To integrate a legacy voice mail system with ShoreTel, you need to perform the following basic tasks:

Configure the server's COM port for SMDI connections to the legacy system.
Configure interface options from ShoreWare Director.
Create a user group for users with access to the integration extensions.

15.10.3.1 COM Port Setup

To establish the SMDI link between the ShoreTel server and the legacy voice mail system, connect one end of a DB-9 serial cable to the COM port on the ShoreTel server and the other end of the cable to a COM port on the legacy voice mail server.

The COM port settings on the ShoreTel server must match the settings of the COM port on the legacy voice mail server. Obtain the legacy voice mail COM port settings from the legacy voice mail server's administration guide or from your system integration manager. You need the following information:

Baud rate
Data bits
Parity
Stop bits
Flow control

To configure COM port communication:

Step 1 From the **Start** menu on the Windows server connected to the legacy voice mail server, select **Settings**, and then **Control Panel**.

Step 2 In the **Control Panel**, open the **Computer Management** folder.

Step 3 Open the **Device Manager**.

Step 4 From the right pane in the window, expand the item **Ports (COM & LTP)**.

Step 5 Right-click the COM port used to connect the ShoreTel server and legacy voice mail system, and select **Properties** from the menu.

Ask your server administrator if you need help in determining the correct COM port.

Step 6 In the **Properties** window, enter the settings for the legacy voice mail server COM port.

Step 7 Click **OK** to save the settings.

Step 8 In ShoreWare Director, open the ***Server*** edit page.

Step 9 Enter the COM port the server will use for SMDI communications in the ***COM Port (1-10)*** text box.

Step 10 Click *Save*.

The ShoreTel system will not read the COM port settings until you have saved the changes to the Server edit page or until the voice mail service is restarted.

15.10.3.2 Analog Trunk Port Setup

The ShoreTel system sends calls to the legacy voice mail server over analog trunks connecting the two systems. The extensions are on the ShoreTel side, and the legacy voice mail system is the trunk side. The ShoreTel system sends calls made to these extensions to the legacy voice mail system when voice mail is needed. Before the call is sent, the SMDI protocol sends information about the call to the legacy voice mail system via the SMDI serial link. This allows the legacy voice mail system to handle the call correctly.

To configure the extensions, you need to do the following:

Create a list of the extensions and include the Logical Terminal Number for each extension.

Configure the extensions with a new dial number (DN) type and marked as private users with no mail box.

Assign a physical port to each extension in Director. Configure the extensions to forward to the Backup Auto Attendant on “no answer” or “busy.”

15.10.3.3 Configuring the ShoreWare Server

Follow these steps to set up communication between ShoreWare Director and the legacy voice mail server.

To set up ShoreWare Director to communicate with the legacy voice mail server:

Step 1 From **ShoreWare Director**, click **Servers** in the navigation frame.

Step 2 Select the server connected to the legacy voice mail system.

Step 3 In the Edit Server page under **Simplified Message Desk Interface**, change the settings as follows:

Step a Make sure that the **ShoreTel as PBX** box is selected.

Step b In the **COM Port** field, enter the port on the server that will be used for SMDI communication.

Step c In the **Message Desk Number** field, enter the Message Desk number (range is 1-999, with a default of 1). This number identifies a specific voice mail system and must be set to the value the voice mail system expects. In configurations where a number of SMDI links are daisy chained together, this value is used to allow each system to known what data belongs to it. Since most systems use only one SMDI link, this parameter is normally set to 1.

Step d In the **Number of Digits** field, enter the extension length. (range 2-32 digits). This value is used to determine how many digits the ShoreTel system sends in SMDI extension fields. This value needs to be set to the value the voice mail system expects. The most common values are either 7 or 10. If the system extension length is less than the number of SMDI digits then the extension number will be padded. For example, if the ShoreTel system needs to send extension 456 and the number of SMDI digits is set to 7, extension

0000456 is sent. If no padding is desired, the number of digits should be set to 2. In the above example with the number of SMDI digits set to 2 only 456 will be sent.

Step e In the Translation Table field, select a translation table. Translation tables are created in ShoreWare Director. If you are using a translation table, make sure the **Use for Call Data** and **Use for MWI Data** check boxes are selected. For more information on building translation tables, see the ***ShoreTel Administration Guide***.

Step f Click **Save**.

15.10.3.4 Digit Translation

If ShoreTel system extensions and legacy voice mail system extensions differ in length, you need to create digit translation tables that map the ShoreTel extensions to legacy system extensions. The digit translation tables must be added as a group of named tables from the Voice Mail section of ShoreWare Director. For more information see the ***ShoreTel Administration Guide***.

Table 15-3 shows a digit translation table mapping shorter ShoreTel extensions to longer legacy system extensions. For example, ShoreTel extensions in the range of 5xx will be in the 65xx range on the PBX, and the original digit “5” will be replaced by “65.”

Extension Mapping		**Digit Translation Table**	
ShoreTel	Legacy	Original Digits	Replacement Digits
5xx	65xx	5	65
3xx	73xx	3	73
2xx	83xx	2	83

Table 15-3 Digit Translation Mapping

Table 15-4 shows a digit translation table mapping longer ShoreTel extensions to shorter legacy system extensions. For example, ShoreTel extensions in the range of 75xx will be in sent. to extensions in the 3xx range on the legacy voice mail system, and the original digit “75” will be replaced by “3.”

Extension Mapping		**Digit Translation Table**	
ShoreTel	Legacy	Original Digits	Replacement Digits
65	5xx	65	5
66xx	6xx	66	6
75xx	3xx	75	3

Table 15-4 Digit Translation Mapping

Figure 15-3 illustrates how digit translation functions between the ShoreTel server and legacy voice system.

To create a digit translation table, follow the procedure below:

Step 1 Launch ShoreWare 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 **System Parameters** link to expand the list.

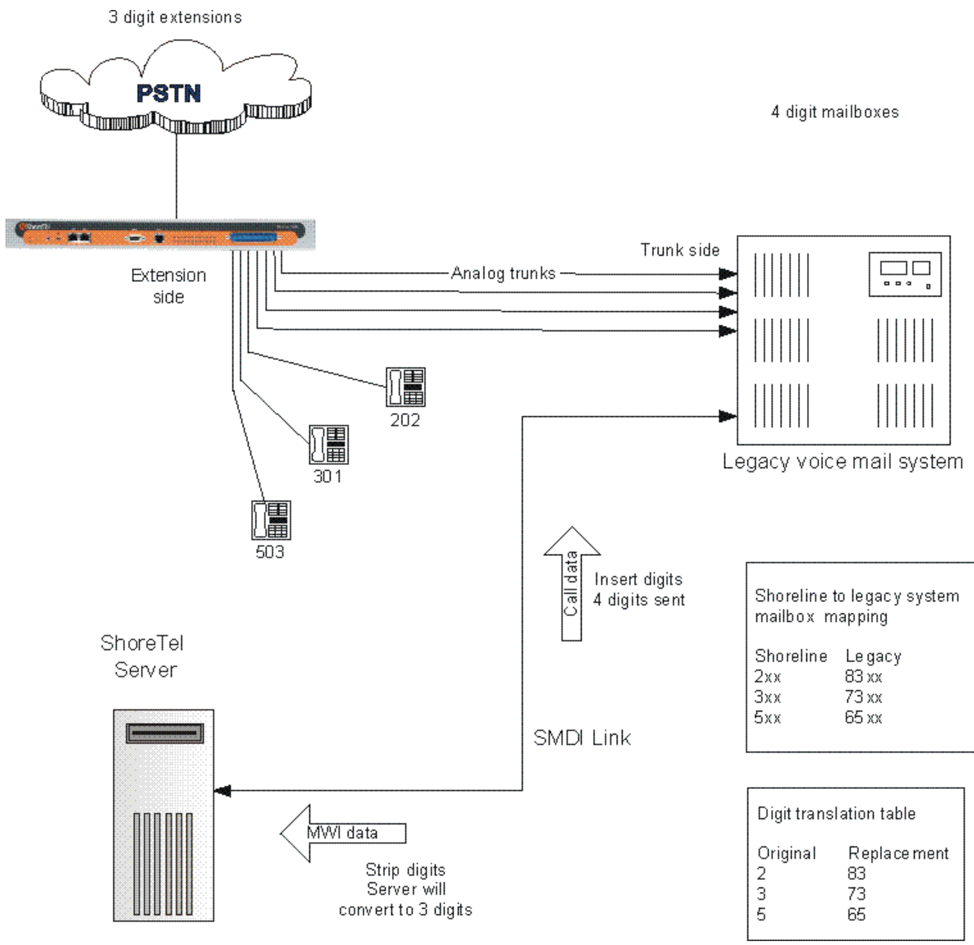


Figure 15-3 Mixed Extension Length SMDI Integration

Step 4 Click on the **Digit Translation Tables** link.

Step 5 Click the **New** button.

Step 6 Enter a name in the **Name** field and click the **Save** button to store your digit translation table.

Step 7 Click the **New** button again to display the **Digit Translation** window (below).

Next, you must select the digit translation mapping that you just created at the server.

Step 8 Click on the **Application Servers** link and click on the name of the ShoreTel server that will be handling the digit translation.

Step 9 In the Simplified Message Desk Interface section of the Application Servers window, select **ShoreTel Voice Mail** from the **Mode** drop-down menu.

Step 10 The **Translation Table** drop-down menu appears. Click on the arrow-button and select the name of the digit translation table that you just created.

Step 11 Select the **Use for Call Data** check box and **Use for MWI Data** check box by placing a check mark in each one (as shown below). Doing so allows for the digit translation to occur when:

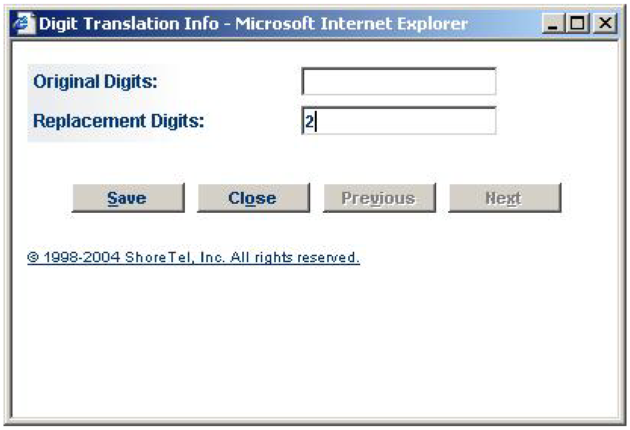


Figure 15-4 Leave Original Digits blank to add a digit to all legacy extensions

Data about a call is transferred between the legacy and ShoreTel systems.

Message Waiting Indicator information is transferred between the two systems to notify the legacy PBX that a message was left on the ShoreTel voice mail.

Step 12 By default, the “Use Flash to Route Calls” check box is enabled. Leave this as is. Note that this check box only appears when “ShoreTel Voice Mail” is selected in the ***Mode*** drop-down menu in the ***Simplified Message Desk Interface*** section of the window. If selected, calls sent to the ShoreTel Auto Attendant from the SMDI trunk group are automatically transferred to the dialed extension using flash. If not selected, calls will be routed using other lines.

The extension length must be the same on each of the systems for the “Transfer Using Flash” feature to work as no translation is applied.

Step 13 Click the **Save** button to store your changes.

15.10.3.5 Setting Up the User Group in ShoreWare Director

Follow these steps to set up a user group for those users who will have their voice mail re-directed to the legacy voice mail system.

To set up the user group:

Step 1 Open **ShoreWare Director**.

Step 2 From the navigation frame, click **Users** and then **User Groups**.

Step 3 Select an existing user group or create a new user group.

Step 4 Change the **Simplified Message Desk Interface Mode** option to **ShoreTel as PBX** by selecting this setting from the drop-down menu.

Step 5 Click **Save**.

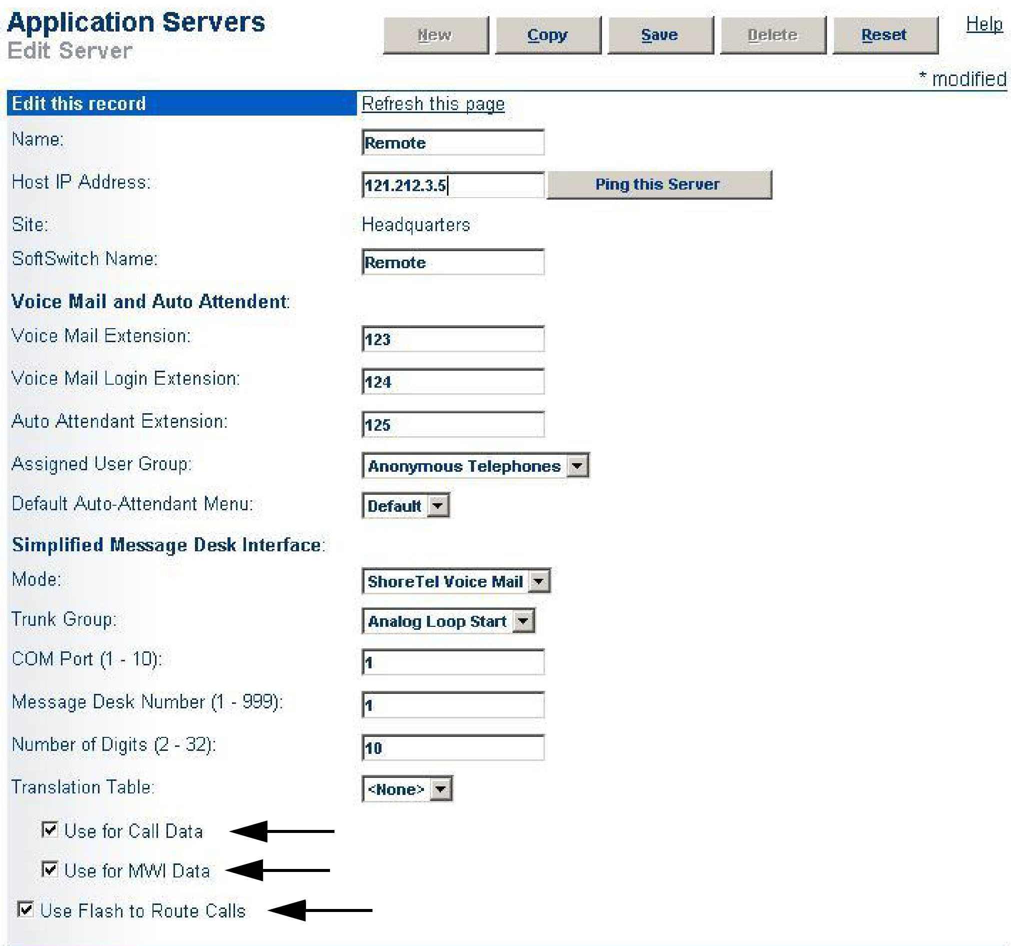


Figure 15-5 Enabling digit translation for MWI and call data, and flash routing

15.10.4 Configuring ShoreTel Voice Mail Integration Using SMDI

As mentioned before, there are two modes of operation with respect to integrating a ShoreTel system and a legacy system:

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

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

The former of these two operational modes (External voice mail) is discussed in Section 15.10.3 on page 208. The procedure for the latter configuration (ShoreTel voice mail) follows.

Configuring the "ShoreTel Voice Mail Configuration" consists of the following major tasks:

Creating a Trunk Group
Creating Trunks
Configuring the ShoreTel Server for SMDI
Creating a User Group
Adding an Individual User
Configuring the Serial Connection
Configuring Digit Translation Tables

PBX link

15.10.4.1 Creating a Trunk Group

One of the first tasks involved in configuring SMDI is to create a trunk group. The trunk group is used to manage the individual trunk lines between the ShoreTel switch and the legacy PBX. Instructions for creating the trunk group are provided below. For additional details on setting up trunk groups, refer to the ***ShoreTel Administration Guide***.

To create a trunk group for SMDI trunks, follow the procedure below:

Step 1 Launch ShoreWare 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 **Trunks** link to expand the list.

Step 4 Click on the **Trunk Groups** link to display the Trunk Groups window.

Step 5 Select the trunk group site, and select ***Analog Loop Start*** for the type. Then click the **Go** link.

Step 6 Enter a name for the trunk group in the **Name** field, as shown below.

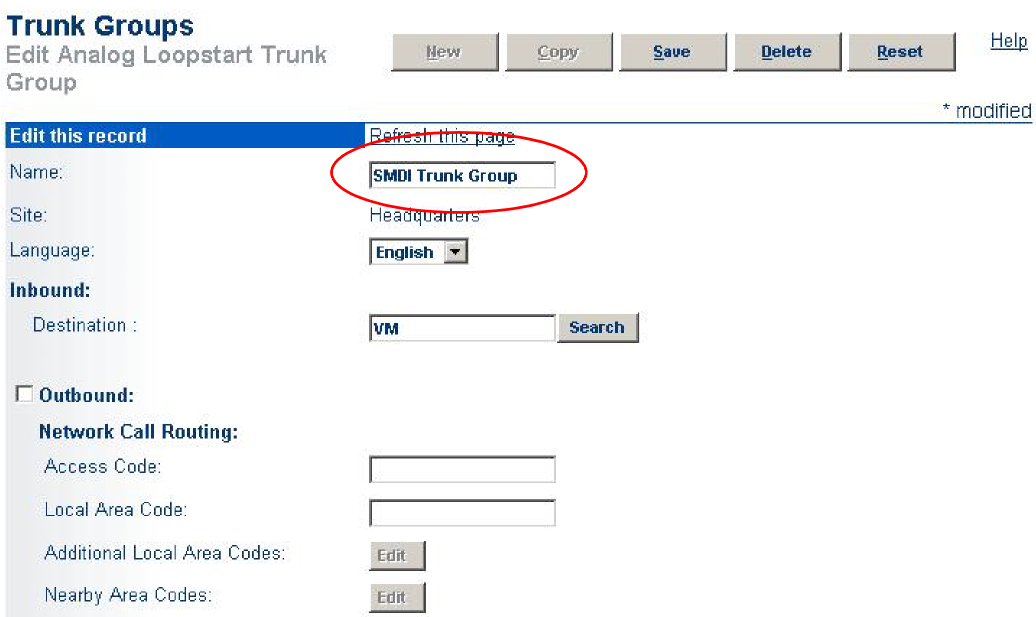


Figure 15-6 Creating a trunk group

Step 7 Enter a voice mail extension in the **Inbound Destination** field to direct inbound calls to the ShoreTel Auto Attendant system.

Step 8 Click the **Save** button to store your changes.

15.10.4.2 Creating Trunks

After creating the trunk group, the next step is to create one or more trunk lines representing each data connection between the ShoreTel switch and the legacy PBX. The lines between the PBX and ShoreTel voice mail must be trunk lines with ShoreTel being the trunk side and the PBX being the extension side, (i.e. calls leaving the PBX for the voice mail system will leave on extensions). The PBX-to-voice mail connection might also be a T1 trunk that uses a channel bank to provide extensions to the legacy PBX.

To create a trunk line, follow the procedure below:

Step 1 With ShoreWare Director still open, click on the **Trunks** link to expand the list.

Step 2 Click on the **Individual Trunks** link.

Step 3 Select the trunk line site (i.e. Headquarters or Remote) from the drop-down menu, and use the drop-down menu to find and select the name of the trunk group you just created.

Step 4 Click the **Go** link to display the Edit Trunk window, similar to the one shown below.

Figure 15-7 Creating a trunk line

Step 5 In the **Number** field, enter the Logical Terminal Number. This value can range from 1 to 9999. For many systems the extension number of the port is used.

The Logical Terminal Number identifies the port the PBX will use to send the call to the ShoreTel voice mail system. It is very important that the LTN match what the PBX will send. You must check with your PBX vendor to determine what will be sent.

Step 6 Click the **Save** button to store your changes.

15.10.4.3 Configuring the ShoreTel Server for SMDI

After creating the trunk lines, you will configure the ShoreTel voice mail server. Configuration involves setting up the various SMDI parameters.

To configure the ShoreTel voice mail server for SMDI operations, follow the procedure below:

Step 1 With ShoreWare Director still open, click on the **Application Servers** link.

Step 2 Click on the name of the server (Headquarters or Remote) that will be acting as the voice mail server for the legacy PBX.

Step 3 In the Simplified Message Desk Interface section of the Application Servers window, click on the drop-down menu and select **ShoreTel Voice Mail**. A new set of fields and menus related to SMDI appear.

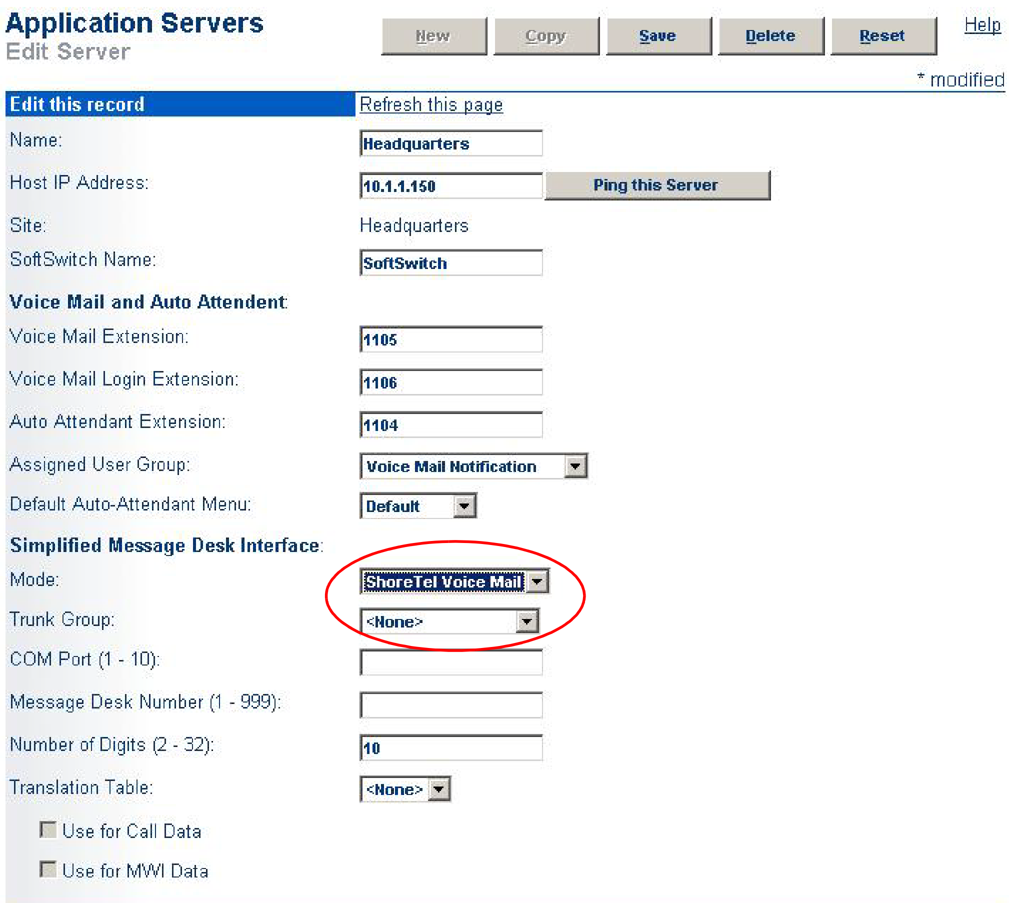


Figure 15-8 **Configuring ShoreTel voice mail server**

Step 4 In the **Trunk Group** drop-down menu, select the name of the SMDI trunk group that you created earlier. This tells the server the name of the trunk group from which it should expect to receive voice mail calls.

Step 5 In the **COM Port** field, enter the numerical value (from 1-10) that corresponds to the serial port of the ShoreTel server where you will be connecting the serial port. (This serial port will be used to route out-of-band SMDI signaling information between the PBX link device and the ShoreTel server.)

Step 6 The Message Desk Number, which has a range of 1-999, is optional and can be set to the default value of 1. Check with the vendor for this value.

The Message Desk Number is used to indicate a specific system in situations where a number of SMDI links have been daisy-chained together. This value allows each system to known which data belongs to it. In most case this parameter is set to 1, since only one system will be using the SMDI link.

Step 7 The Number of Digits field, which has a range of 2-32, is optional.

This value determines how many digits the ShoreTel system will send in SMDI extension fields. This value needs to be set to the value the voice mail system expects. The most common values are either 7 or 10. If the system extension length is less than the number of SMDI digits, then the extension number will be padded. For example, if the ShoreTel system needs to send extension 456 and the number of SMDI digits is set to 7, extension 0000456 will be sent. If no padding is desired the number of digits should be set to 2. In the above example with the number of SMDI digits set to 2 only 456 will be sent.

Step 8 The translation table is optional and can be left as is for now. We will be returning to the related topic of digit translation tables later.

Step 9 Click the **Save** button to store your changes.

15.10.4.4 Creating a User Group

After setting up the ShoreTel voice mail server for SMDI, the next step is to add users to the system. You will create a user group, and in this user group you will specify that all members will use ShoreTel Voice Mail. Once this is done, then you will modify user profiles at the individual level. For now, we will talk about creating the user group.

To create a user group for users on the legacy PBX system, follow the procedure below:

Step 1 With ShoreWare Director still open, click on the **Users** link to expand the list.

Step a Click on the **User Groups** link.

Step b Click on the **Add New** link to display the **User Groups** window.

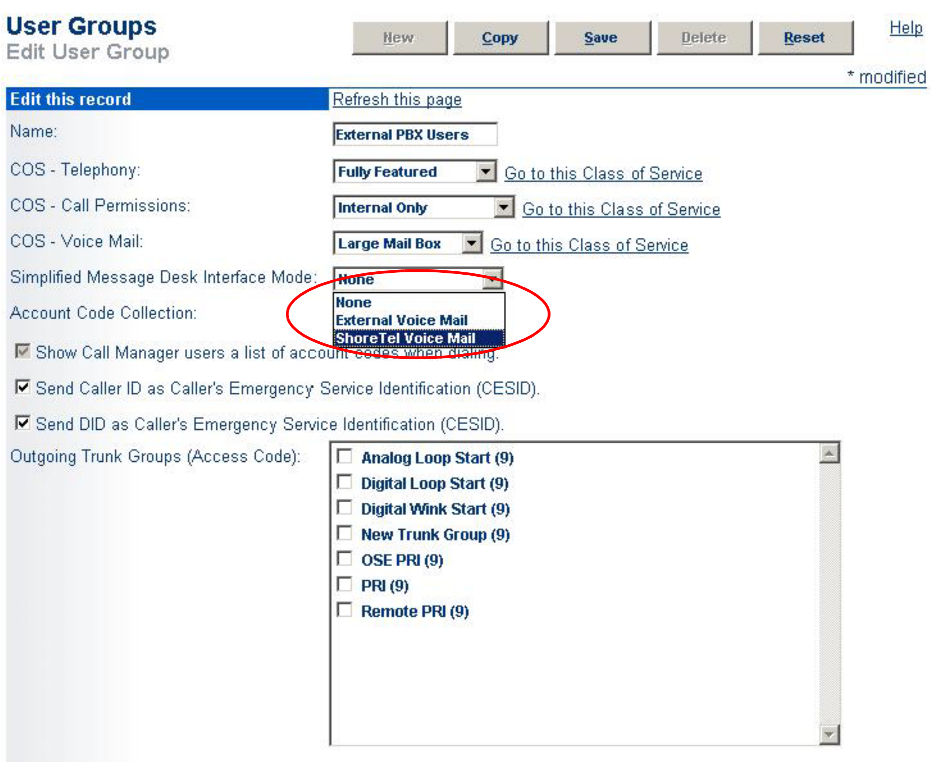


Figure 15-9 **Creating a user group for legacy users**

Step 2 Enter a Name for the user group in the **Name** field.

Step 3 In the Simplified Message Desk Interface Mode drop-down window, select **ShoreTel Voice Mail** from the list.

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

15.10.4.5 Adding an Individual User

After creating the user group, you can create user profiles for the legacy PBX users. To do so, follow the procedure below:

Step 1 With ShoreWare Director still open, click on the **Users** link to expand the list.

Step 2 Click on the **Individual Users** link.

Step 3 In the **Add new user at site** field, select the server where you configured the ShoreTel voice mail for the PBX link device.

Step 4 Click the **Go** link to display the **Edit User** window, shown below.

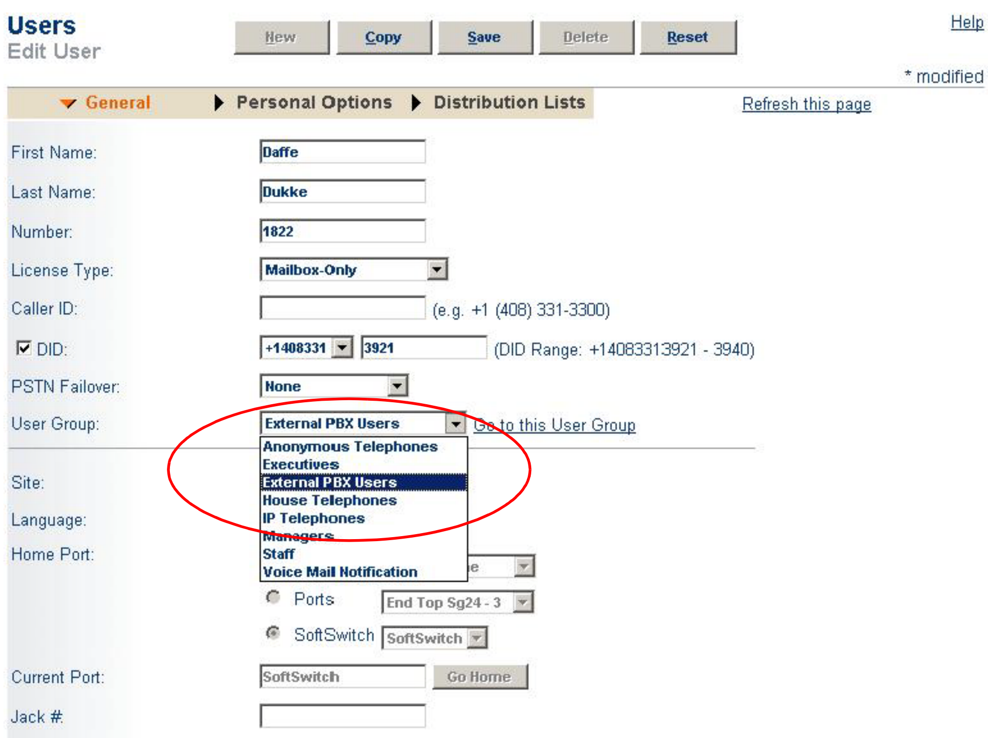


Figure 15-10 Creating a user record for a legacy user

Step 5 Enter a name for the user in the **First Name** and **Last Name** fields.

Step 6 In the License Type drop-down menu, click on the arrow-button and select **Mailbox-Only**. The user is located on the legacy system and thus, he or she does not require a ShoreTel extension.

Step 7 In the User Group drop-down menu, click on the arrow-button and find and select the name of the user group you just created.

Step 8 Click the **Save** button to store your changes.

15.10.4.6 Configuring the Serial Connection

The ShoreTel voice mail system will only support one serial link per application server. To support another legacy PBX, you will need another ShoreTel distributed application server. A serial cable (i.e. null modem) should be used to connect the legacy PBX to one of the COM ports of the ShoreTel server. Note that the ShoreTel system will extract the serial port settings, such as baud rate and parity bit values, from the Windows COM port settings. These settings can be verified by following the procedure below:

Step 1 Right-click **My Computer**.

Step 2 Select **Manage**.

Step 3 Select **Device Manager**.

Step 4 Left-click on **Ports (COM & LPT)**.

Step 5 Right-click **Communications Port (COM1)**, and select **Properties**.

Step 6 Left-click on the **Port Settings** tab.

Step 7 Verify that the settings match those suggested by the documentation that came with your legacy PBX device.

15.10.4.7 PBX

Manufacturer	Model
Nortel	• Meridian 1 • Nortel Norstar
Avaya	• System 75/85 • Definity
Mitel	• SX50 • SX200 • SX2000
Siemens	• 300S
NEC	• NEAX

Table 15-5 Supported PBXs

15.10.4.8 PBX link

A PBXLink device may be needed to provide SMDI services for a legacy PBX that does not offer support for SMDI. The PBXLink devices, manufactured by CTL, provides integration services to allow certain digital PBXs to interface seamlessly with a Voice Messaging System. The PBXLink connects to the PBX using a digital telephone line and to the Voice Messaging System using an RS-232 link. The PBXLink uses information appearing on the emulated digital set to determine the original source and destination of the calls being forwarded to the voice mail system. This information is then communicated to the voice mail system on an RS-232 serial link using the industry standard “Centrex SMDI” protocol. The PBXLink is compatible with SMDI-compatible voice mail systems.

When using SMDI, ShoreTel voice mail configuration, the following features will not be supported:

Extension Assignment
Setting call handling mode
Setting agent state

The following features will be supported:

Recording greeting and name
Setting TUI password
Enable/disable envelope information
Email voice message options
Find Me
Message functions including call back
Message sending functions
Workgroup
ShoreTel voice mail
Agents cannot be extensions in the legacy PBX
System configuration
Configuration parameters

15.11 System Requirements

The following are required on the ShoreTel system, or on the legacy PBX to enable the integration of the two systems:

ShoreTel system

— ShoreGear Voice Switch that supports a T1 circuit.

Legacy PBX

— T1 or PRI card for the PBX
— Available card slot and capacity for the added trunks
— Required software or licenses to support the desired trunk interface

If PRI is used in the integration interface, the legacy PBX must emulate the CO or support Network Side PRI.

15.12 Connection Cable

15.12.1 Special Considerations - Nortel PBX

When integrating with a Nortel Meridian PBX, a T1 connection must be used since the legacy system does not support Network Side PRI.

15.12.2 Special Considerations - Avaya/Lucent PBX

Universal Dial Plan (UDP) Must be Active - This capability enables transparent dialing between the Avaya/Lucent PBX and the ShoreTel system. If this is not active, users on the PBX will either have to dial a trunk access code to reach the users on the ShoreTel system, or configure forwarding from an extension in the legacy system to the ShoreTel extension using the trunk access code and the extension.

In some cases, this feature must be purchased separately from Avaya/Lucent.

15.13 Administration and Configuration

15.13.1 Tie Trunk Configuration

The following summary describes the administration and configuration of the digital trunk for connecting the ShoreTel system to the legacy system.

15.13.2 Services Summary

Before starting, a summary of the required configuration should be made based on the required services in the interface.

Desired Service	Required Configuration
Extension-to-Extension Calling	Enable inbound services on the trunk. Direct inbound calls using extension routing to the ShoreTel extensions. Enable off-system extensions. Define off-system extension range to match extensions on the remote PBX.
Inbound Trunks on Remote PBX	Enable inbound services on the trunk. Direct inbound calls using extension routing to the ShoreTel extensions. Outbound trunks on the remote PBX enable outbound services on the trunk. Configure any required access code for the trunk and the local area code for the trunks connected to the remote PBX. Configure the desired trunk services such as local, long distance, and so on. Configure the dialing format and any required digit sequences that are to be pre-pended to the dialed numbers. Users require trunk group access rights to use the trunk for outbound calls.
Consolidated Long Distance	Enable outbound services on the trunk. Configure any required access code for the trunk and the local area code for the trunks connected to the remote PBX. Configure trunk services, such as long distance and international. Configure the dialing format and any required digit sequences that are to be pre-pended to the dialed numbers. Users require trunk group access rights to use the trunk for outbound calls.

Table 15-6 Service Configuration Requirements

15.14 Trunk Configuration

The following steps describe how to configure the trunk for integrating the legacy PBX and the ShoreTel system. Some steps are optional depending on the types of services desired as summarized above.

To create a new trunk group

Step 1 In the ShoreWare Director, select ***Trunk Groups*** from the navigation frame to open the **Trunk Groups list** page.

Step 2 Select the site where the trunk will be integrated and the type of trunk to configure - Digital Wink Start for T1 or PRI for PRI - and select ***Go***. The new trunk group is created and the **Trunk Group Edit** page appears.

Step 3 Click **Save** to store the trunk group configuration changes.

To configure inbound services with extension routing

Step 1 In ShoreWare Director, open the ***Trunk Group*** edit page for the tie trunk.

Step 2 Configure the number of digits received to match the number of digits sent by the remote PBX. This must match the extension length.

Step 3 Enable ***Extension Routing*** by checking the box. This directs all the received calls to the configured ShoreTel extension that matches the received DNIS digits.

Step 4 Select a ***Destination*** to provide a back-up when the received digits do not match an extension in the ShoreTel system.

Step 5 Click ***Save*** to save the trunk group configuration.

To configure off-system extensions

Step 1 In ShoreWare Director, open the ***Trunk Group*** edit page for the tie trunk.

Step 2 Select the ***Edit*** button by the off-system Extensions. The ***Off Systems Extension Range*** dialog is displayed.

Step 3 Click ***New*** and define the extension ranges for the extension off the remote PBX.

Step 4 Click ***Save*** to save the trunk group configuration.

To configure outbound call routing (via the remote PBX)

Step 1 In ShoreWare Director, open the ***Trunk Group*** edit Page for the tie trunk.

Step 2 Enable outbound services by selecting the ***Outbound*** check box.

Step 3 Configure the access code and areas codes for the trunk to match the PSTN connection of the remote PBX.

Step 4 Select the desired trunk services to match the services provided via the remote PBX.

Step 5 Select the desired ***Trunk Digit Manipulations*** to match the tie trunk and the required dialing for the PSTN connection to your legacy PBX.

Step 6 As needed, configure the local prefixes and pre-pend digits to match the tie trunk and the required dialing for the PSTN connection to your legacy PBX.

For additional information on trunk configuration and information on configuration options, refer to the ***ShoreTel Administration Guide***.

CHAPTER 16

IP Phone Installation

With ShoreTel IP phones, you deploy your telephony system as an end-to-end IP network without dedicated station wiring. Connecting anywhere on the network, ShoreTel IP phones work with the ShoreTel Communicator applications or can be used independently, providing an intuitive interface to essential telephone features.

ShoreTel IP phones are preconfigured by ShoreTel to work in conjunction with your ShoreTel system and your network's Dynamic Host Configuration Protocol (DHCP) server. Once the servers are configured, you simply plug the phones into the network and they are automatically added to your ShoreTel system.

If you are not using a DHCP server or it is not currently online, you can set a static IP address and other startup parameters directly at the IP phone. See Appendix C, starting on page 271, for more information.

16.1 Checklist

Review the following IP phone installation topics before proceeding to the next chapter:

Task	Description
Recommendations	page 225
Preparing Your ShoreTel System for IP Phones	page 226
Associating a User Group with Unassigned IP Phones	page 229

Table 16-1 IP Phone Installation Checklist

16.2 Recommendations

The following recommendations will help you install your IP phones.

Make sure you have reviewed your network bandwidth and Quality of Service (QoS) strategies and configured your network for your IP phones as described in Chapter 9, starting on page 107.

Make sure you have configured DHCP vendor option 156 with boot server information.

The phones may not boot properly if incorrect configuration data is present in the telephone. This can occur if the telephones were previously used in an environment where DHCP and automatic provisioning was not used, or the telephone is from a vendor other than ShoreTel. See Appendix C, starting on page 271, for information about changing the telephone to the correct settings.

16.3 Preparing Your ShoreTel System for IP Phones

This section provides the information you need to prepare your ShoreTel system for IP phones.

16.3.1 Configuring Voice Switches for IP Phone Support

To provide PSTN local dialing for IP phone users, every site where IP phones are in use must have a ShoreGear switch configured to support the number of IP phones at the site, plus local analog or T1 trunks.

The ShoreGear voice switches send a heartbeat to the IP phones every 60 seconds. If the heartbeat is not acknowledged within approximately four seconds, the switch considers the IP phone to be offline or unavailable. The voice switches continue to broadcast the heatbeat every minute. Any currently-offline IP phone that returns an acknowledgement is considered online and available.

To configure IP phone support on a ShoreGear voice switch, you must reserve ports for IP phone support on the ShoreGear ***Switch*** edit page in the ShoreWare Director. See the “Configuring Switches” chapter in the ***ShoreTel Administration Guide*** for additional information.

16.3.2 Configuring Teleworker IP Phones

To configure an IP phone as a teleworker phone:

Step 1 Define a range of IP addresses set aside for IP phone teleworkers as described in Section 16.3.4 on page 227.

Step 2 Set a static IP address for the IP phone included in the range you defined in Step 1. For instructions on setting a static IP address for an IP phone, see Appendix C, starting on page 271.

Step 3 Connect the IP phone to your Ethernet connection to the Internet.

16.3.3 Assigning the Configuration Switches

You need to designate a switch for handling initial service requests from IP phones installed on your ShoreTel system. You have the option of assigning two switches to this function, to provide a backup in case of network problems. Every IP phone installation must have at least one configuration switch. If you do not assign a switch, the ShoreTel system automatically assigns the first two ShoreGear switches that you configure.

IP phones must be able to contact at least one of the assigned configuration switches when first connected to the network. If the IP phone cannot reach a configuration switch, the telephone will not be added to the system.

To assign configuration switches:

Step 1 From the ShoreWare Director navigation pane, click **IP Phones**.

Step 2 Click **IP Phones Options**. Figure 16-2 shows the **IP Phones Options** edit page. This page has several configurable parameters:

IP Phone Configuration Switch 1
IP Phone Configuration Switch 2

User Group for Unassigned IP Phones
IP Phone Announcement
IP Phone Password
Enable IP Phone Failover
Delay After Collecting Digits

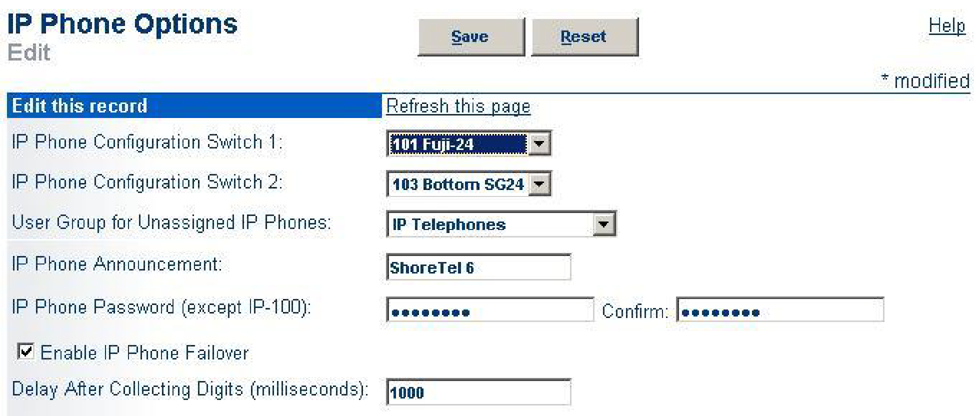


Figure 16-1 IP Phones Options Edit Page

Step 3 Select an available switch from the pull-down lists for configuration switches 1 and 2.

For information on the other IP phone options, see the ***ShoreTel Administration Guide***.

16.3.4 Setting IP Address Ranges

If your system consists of more than one site, you must define an IP address range for IP phones at each site in the system. Setting ranges for each site ensures that new phones added to the system will be associated with the correct voice switch at the telephone's site.

You can view the IP address range for each site from the **IP Address Map** list page, shown in Figure 16-2. The page lists the sites and associated IP address ranges.

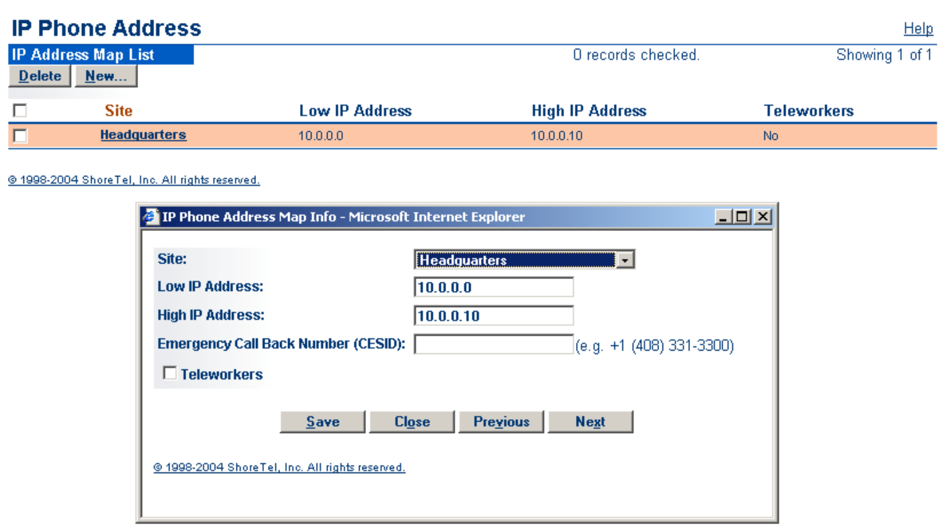


Figure 16-2 IP Address Map List and IP Address Range Edit Pages

To add a site with IP phones, click **New** and enter the information on the **Site IP Address Range** edit page (see Figure 16-2). To delete a site from the list, click the check box to the left of the site and click the **Delete** button.

To edit the IP address range for a site:

Step 1 On the IP Address Map List page, in the ***Site*** column, click the site for which you are setting a range. The **Site IP Address Range** edit page appears as shown in Figure 16-2.

Step 2 If you are setting the IP address range for a site other than shown in the ***Site*** field, select it from the list.

Step 3 Enter the lowest IP address in the **Low IP Address** field.

Step 4 Enter the highest IP address in the **High IP Address** field.

Step 5 If you are setting a range for teleworker IP phones, click the **Teleworkers** check box.

Step 6 To set the new range, click **Save**. You can set ranges for other sites in the system by clicking **Previous** or **Next**.

If a phone is added with an address that is not within a specified range for any site, or there are no IP address ranges defined for any site, the telephone will be automatically assigned to the headquarters site. This causes seven-digit numbers dialed from the IP phone to be dialed as numbers within the area code of the headquarters site. In addition, this causes all telephone calls to users who are not at the headquarters to use the configured inter-site voice encoding for that system.

16.3.5 802.x Authentication

ShoreTel IP Phones have supported 802.1x network authentication since ST 9. This authentication requires the device to present a userid and a password. 802.1x is enabled by default. The default SID (userid) of the last 6 characters of the MAC address of the phone.The passwordmust be entered manually (no default) the first time the phone boots andis then cached if authentication succeeds.

If 802.1x enabled on the phone and the network is not setup to handle the feature, the phone will boot as normal.

If upgrading from another firmware that supports 802.1x (3.3.x or 3.4.x), the previous settings (802.1x on/off, SID, password) will be preserved. If upgrading from a firmware that does not support 802.1x (2.2, 2.3, 3.1, 3.2) Logical Link Discovery Protocol (LLDP)will be turned on by default anda default SID of the last 6 characters of the MAC address will be applied.

16.3.6 DHCP Settings

ShoreTel IP phones are preconfigured to use the network's DHCP server for addressing. In addition to its address and standard network addresses, the DHCP server's response also provides the following:

ShoreTel server address: The ShoreTel server's address is used to access and download the latest telephone application software and the configuration information for the ShoreTel system.

SNTP server: The SNTP provides a standard network time to maintain the telephone's displayed time and date.

16.3.6.1 ShoreTel Server Address

The ShoreTel server provides the IP phones with the latest application software and the configuration information that enables the IP phone to be automatically added to the ShoreTel system. The ShoreTel server's address must be provided to the phone as a vendor-specific option. ShorePhones are preconfigured to look for the ShoreTel server's address to be specified as Vendor Specific DHCP Option 156. If these options are not available, the ShoreTel IP phones will use Option 66.

For help on configuring these DHCP Options, see Section 9.7 on page 123.

16.3.6.2 SNTP Server

The DHCP server should be configured to provided the address of your network's SNTP server to provide date information to the IP phones.

16.4 Associating a User Group with Unassigned IP Phones

Unassigned IP phones are available for users configured for Any IP Phone. Select the user group that will have access to unassigned IP phones from the pull-down list.

Since unassigned IP phones are not associated with a user, you cannot report on calls made from these telephones and associate them with an individual user. It is recommended that unassigned IP phones be configured with a class of service with minimal calling privileges.

CHAPTER 17

Server Installation

This chapter describes installation procedures for main and distributed ShoreWare servers. Review the following server installation topics before moving on to the next chapter:

Task
Installing Software on the Main Server
Installing Software on a Distributed Server
Ensuring Proper Server Performance

Table 17-1 Server Installation Checklist

17.1 Installing ShoreTel Software

Before beginning software installation, close all programs and verify that no anti-virus software is running.

Install the ShoreWare server onto an NTFS partition. Do not install the ShoreWare server software onto a FAT partition, especially the ShoreTel data folder. FAT partitions are restricted to 16-bit DOS addressing methods, which limit the size of the partition to 2 GB (insufficient for the ShoreTel application).

Prior to installing the software, verify that the Data Execution Prevention settings have been set correctly.

Installing ShoreTel Server Software Basic Concepts:

The default parameters presented by the ShoreTel installer are recommended. However, if ShoreTel software is to be located in a different location, select the correct installation path during the install process.

ShoreTel Server Setup checks for prerequisite software. If the required software is not installed, setup will automatically stop and it will be necessary to install the proper prerequisite software before continuing.

When the Install Shield Wizard has completed, you will be prompted to restart your server. Click “Finish” to restart.

Once the server has restarted, you may be prompted to configure a TAPI service provider. Enter the appropriate access and area codes and then continue. The system will need to be restarted. It will typically take 30 to 60 seconds after the operating system is up and running for the Microsoft Internet Information Services (IIS) and the ShoreTel services to be functional.

Launch ShoreTel Director by clicking the ShoreTel Director desktop icon. If IIS is not yet running, an error will be displayed.

If this is the first time you are logging into ShoreTel Director, use the default user ID and password of “admin” and “changeme.” You will also need to register your product.

Once you have successfully logged into ShoreTel Director, you will be taken to the QuickLook maintenance page. You should confirm that all ShoreTel services are running.

17.1.1 Upgrading ShoreTel Server Software

If you are upgrading your ShoreWare Headquarters server, follow the same process used for installing new software. Setup will automatically determine that an upgrade is in process, and you will be presented with a subset of the installation wizard screens. (There is no need to change the destination folders of the ShoreWare files.)

Setup will look for the ShoreTel database. If a database is found and it is an older version, Setup will make a backup copy and convert the database to the latest release. Note that Setup will not overwrite an existing database.

All voice applications (voice mail, automated attendant, workgroups, and so on) are affected until the upgrade is complete.

After the installation, a panel warns that the installation will stop all ShoreWare services (Figure 17-1).

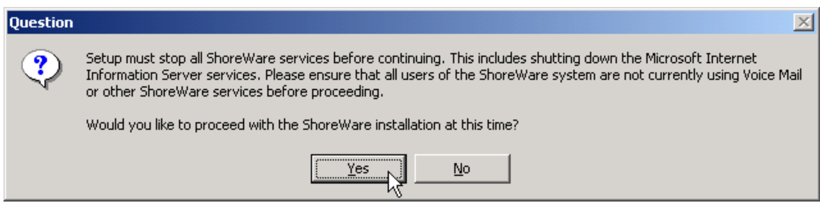


Figure 17-1 Warning

To finish the upgrade, restart your ShoreGear voice switches so that they will upgrade their firmware (this affects all calls in progress). Then upgrade your distributed servers.

The Distributed ShoreWare server must be a **dedicated** server with **no** other applications installed. This means you should **not** use this server for any of the following: Windows Domain controller, Terminal Server, Database Server (with MySQL), Web server, nor exchange server. This DVS server must be exclusively dedicated to supporting ShoreWare.

17.2 Ensuring Proper Server Performance

The following are some guidelines for ensuring the best performance from your ShoreWare server. This by no means is an exhaustive list. Please refer to a reference book on the subject or information on the web at www.microsoft.com.

Verify the server meets the hardware requirements, especially memory.

Make sure the hard disk is not fragmented.

Make sure you optimize server performance for background services rather than for applications. The voice services running on the server are real-time services that could be negatively affected by having an application running in the foreground.

— To configure this option, go to Control Panel and open the **System** icon. In the System Properties window (Figure 17-2), click the **Advanced** tab and then click the **Performance Options** button. From the Performance Options window, select the option to optimize performance for **Background services**.

Make sure the paging file size (virtual memory) on the server is large enough.

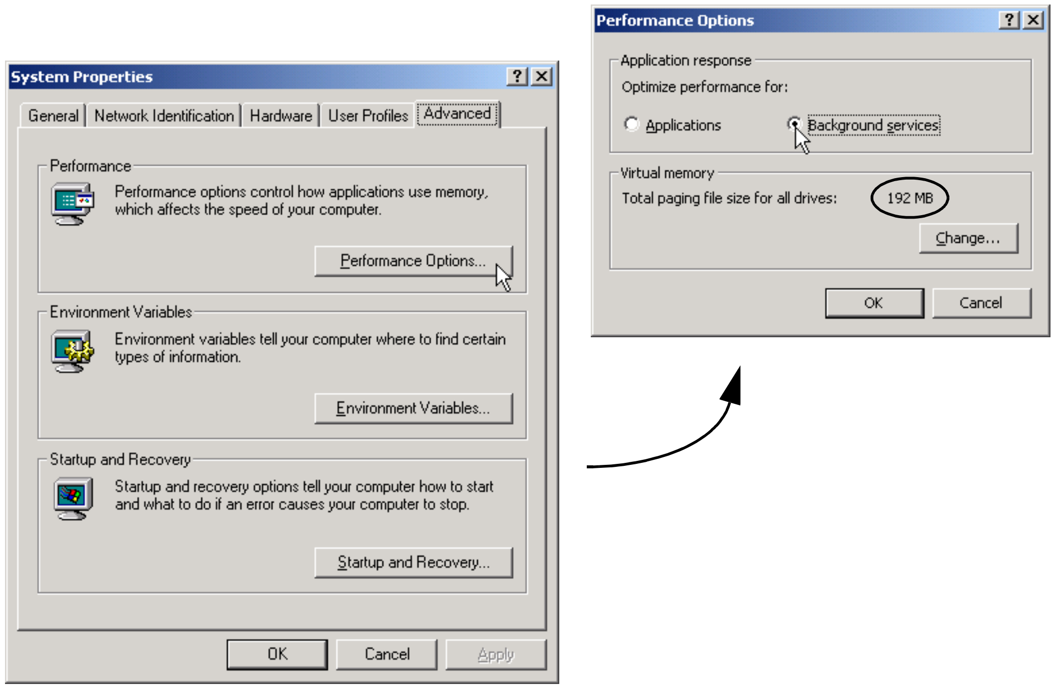


Figure 17-2 System Control Panel and Performance Options

— To check the paging file size, go back to the Performance Options window shown in Figure 17-2. The paging file size should be 1 to 3 times larger than the physical memory on the server. If you have 512 MB of memory, the paging file size should be between 512 MB and 1536 MB. Increase the paging file size by clicking the **Change** button.

Make sure you set the server to maximize for network performance.

— To configure this option, go to Control Panel, open the **Network and Dial-up Connections** icon, and then open the **Local Area Connection** icon. From the Local Area Connection Properties window, select the **File and Printer Sharing for Microsoft Networks** item and click **Properties**.

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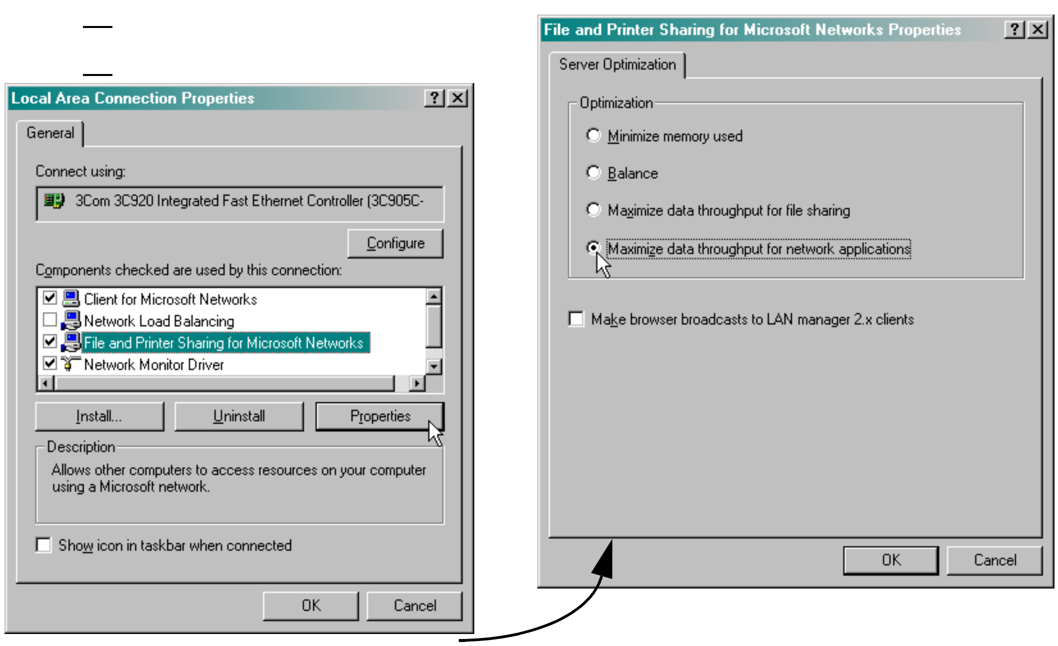

17.3 Upgrading ShoreTel Servers from Windows 2003 (32-bit) to Windows 2008 (32-bit)

The following procedure is required when upgrading the operating system to Windows Server 2008 (32-bit) on a server that is running ShoreTel server software.

Warning: Active Directory settings must be disabled before proceeding with the upgrade process.If Active Directory is not disabled, you will not be able to start ShoreWare Director after upgrading.

NOTE: When installing ShoreWare 11x on Windows 2008 (32-bit) you must launch Setup.exe using "Run as Administrator."

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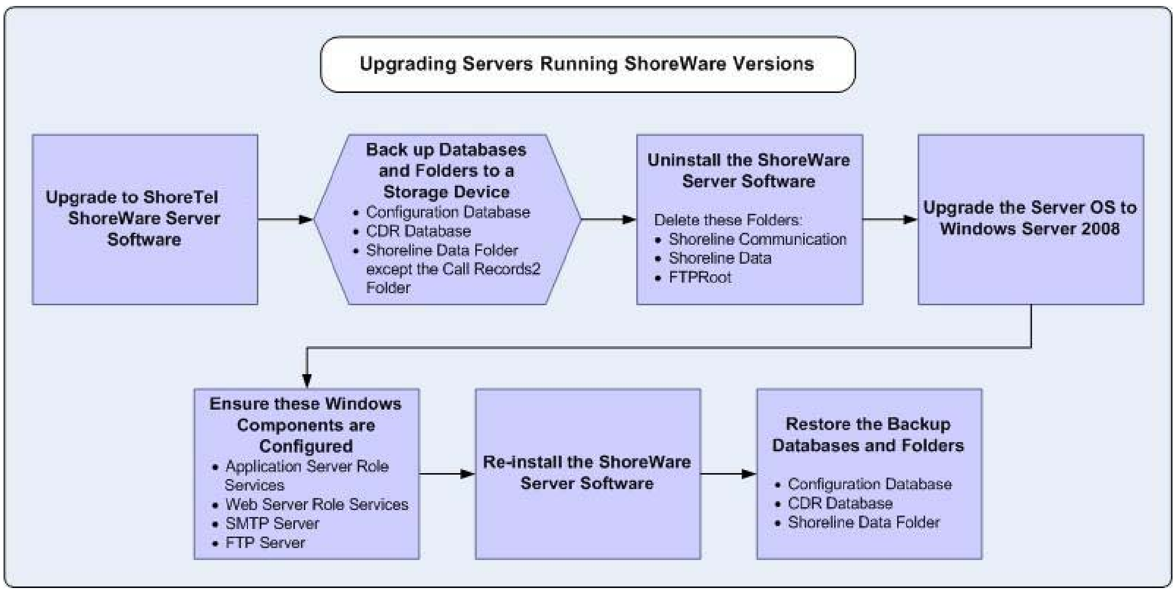


Figure 17-3 Upgrading Servers Running Previous ShoreWare Software

Step 1 Upgrade the ShoreTel Server software to Version 11.

Step 2 Disable Active Directory if you have this option enabled.

Step 3 Stop all ShoreTel services and backup the Configuration and CDR databases

Refer to the ShoreTel Administrator Guide for instructions on backing up and restoring a ShoreTel database.

NOTE: ShoreTel recommends that you back up these databases to a storage device separate from the server you intend to upgrade.

Step 4 Uninstall the ShoreTel Server software and delete the following folders:

Shoreline Data

Shoreline Communications

FTPRoot

NOTE: Make sure Windows PowerShell is not present. Refer to Microsoft support for information on uninstalling Windows PowerShell.

Step 5 .Upgrade the server operating system to Windows Server 2008 (32-bit). After the operating system is upgraded, you must activate the operating system prior to installing ShoreTel Server software.

Do nut use Autorun from the Windows Server 2008 CD. Autorun will only allow a new installation of Windows 2008.

Warining: If you did not disable Active Direcotry prior to upgrading to Windows 2008 and Active Directory is then enabled after upgrading, you will not be able to access ShoreTel Director.

Step 6 Ensure that the Application Server Roles and Web Server Roles are configured.

Step 7 Install the ShoreTel Server software.

Step 8 Restore the Configuration and CDR database files.

Refer to the ShoreTel Administrator Guide for instruction on restoring a backup copy of a ShoreTel database

NOTE: Active Directory Login will not be available after upgrading to Windows 2008.

17.3.1 Existing HQ server hardware will not support the 64-bit version of Windows Server 2008 (R2)

Step 1 Disable Active Directory and Distribute d Database, if enabled

Step 2 Backup the existing ShoreTel data files

Step 3 Backup the existing ShoreTel databases

Step 4 Record the IP address, gateway IP address, netmask value and other network parameters of the current server

Step 5 Bring down the current system and Install Windows 2008 R2 on new hardware

Step 6 Set the network values on the new server to match what was recorded from the old server

Step 7 Prepare the new operating system with the necessary components (Application Server Role Services, Web Server Role Services, SMTP Server, etc.

Step 8 Copy the Shoreline Data backup in the proper location

Step 9 Install ShoreTel 11 on the new platform

17.3.2 Existing DVS server hardware will not support the 64-bit version of Windows Server 2008 (R2)

Step 1 Backup the existing ShoreTel data files

Step 2 Disable Distributed Database, if enabled

Step 3 Record the IP address, gateway IP address, netmask value and other network parameters of the current DVS server

Step 4 Bring down the current DVS system and Install Windows 2008 R2 on new hardware

Step 5 Set the network values on the new DVS server to match what was recorded from the old server

Step 6 Prepare the new operating system with the necessary components (Application Server Role Services, Web Server Role Services, SMTP Server, etc.

Step 7 Install ShoreTel 11 on the new platform

17.3.3 Existing HQ Server hardware will be upgraded to Windows Server 2008 R2

Step 1 Disable Active Directoryand Distributed Database, if enabled

Step 2 Backup the existing ShoreTel data files

Step 3 Backup the existing ShoreTel databases

Step 4 Uninstall ShoreTell server software and remove the Shoreline Data and ShoreWare Server folders

Step 5 Record the IP address, gateway IP address, netmask value and other network parameters of the current server

Step 6 Upgrade the server platform with Windows 2008 R2 on the old server platform (complete wipeout installation of Windows 2008 R2)

Step 7 Confirm the network values on the server match what was recorded from the old server

Step 8 Prepare the new operating system with the necessary components (Application Server Role Services, Web Server Role Services, SMTP Server, etc.

Step 9 Copy the Shoreline Data backup into the proper location

Step 10 Install ShoreTel 11 on the new platform

17.3.4 Existing DVS Server hardware will be upgraded to Windows Server 2008 R2

Step 1 Backup the existing ShoreTel data files

Step 2 Disable Distributed Database, if enabled

Step 3 Uninstall ShoreTell server software and remove the Shoreline Data and ShoreWare Server folders

Step 4 Record the IP address, gateway IP address, netmask value and other network parameters of the current server

Step 5 Upgrade the server platform with Windows 2008 R2 on the old server platform (complete wipeout installation of Windows 2008 R2)

Step 6 Confirm the network values on the server match what was recorded from the old server

Step 7 Prepare the new operating system with the necessary components (Application Server Role Services, Web Server Role Services, SMTP Server, etc.

Step 8 Install ShoreTel 11 on the new platform

CHAPTER 18

Desktop Installation

This chapter describes the procedure for installing ShoreTel Communicator on desktop computers. You can install ShoreTel Communicator or have users install ShoreTel Communicator, in which case the server can notify them with information on their extensions and how to install the ShoreTel Communicator.

18.1 Checklist

Review the following installation topics before proceeding to the next chapter:

Task	Description
Recommendations	page 239
Notifying Users via Email	page 240
Installation Procedure	page 240
Installing Outlook Integration	page 246
Upgrade Procedures	page 248
User Licensing	page 248
Other Considerations	page 251

Table 18-1 Desktop Installation Checklist

See Chapter 18, starting on page 239 for all hardware and software requirements for the ShoreWare ShoreTel Communicator application.

18.2 Recommendations

The following recommendations will assist you in installing the ShoreTel Communicator application on your desktop computer.

Verify you have your server name, user name, password, and extension number. These are required when you start the ShoreTel Communicator application for the first time.

Close all applications before starting the ShoreWare software installation.

With the Silent Client Install feature, the client software upgrade process on remote machines do not require administrative rights by the person installing or upgrading software on client machines. Administrators can upgrade the software on all client machines, using Active Directory Group Policies, regardless of the permissions associated with those machines or the users who log into those machines.

Many of the changes are reliant on Microsoft Active Directory. Microsoft Outlook must be configured in "Corporate or Workgroup" mode for Outlook integration to function properly. "Internet Only" mode is not supported.

18.3 Notifying Users via Email

To simplify installation, the ShoreTel system provides an integrated software distribution feature. Using ShoreWare Director, the system administrator can send an email message to each user configured with an email address.

You can send all users, some users, or just one user an email message using the **Notify Users** page (Figure 18-1).

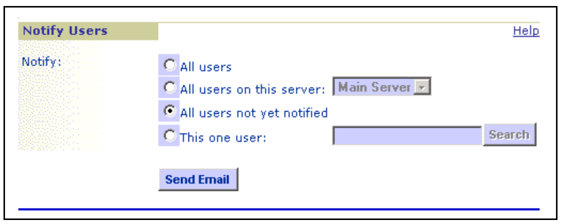


Figure 18-1 Notify Users page

18.4 Installation Procedure

This section provides the most typical steps associated with installing the ShoreWare ShoreTel Communicator application. There are two methods of installing ShoreTel Communicator software:

Silent Client Upgrade
Standard Integrated Software Distribution

18.4.1 Silent Client Upgrade

The Silent Client Upgrade process allows for the upgrading of the client software on remote machines such that administrative rights are no longer needed by the person running the install/upgrade or the client machines. An administrator can easily upgrade the software on all client machines regardless of the permissions associated with those machines or the users who log into those machines.

Many of the changes are reliant on Microsoft Active Directory. The Microsoft Active Directory software handles the following tasks:

Create a Group Policy Object to use to distribute the software package

Assign a package to a group of computers running Windows 2003, or XP Professional, or Vista

Publish a package

Remove a package

You will need to install the following files from the Client DVD with file permissions set to Share and File level Access by group <everyone>:

Data1.cab
Setup.exe
ShoreWare Call Manger.msi

For Microsoft Outlook 2007 Plug-in with Offline Call Handling Modes, the following items must also be pushed separately.

Microsoft Office 2007 Primary Interop Assemblies

Visual Studio Tools for the Office system 3.0 Runtime

Enabling the new Remote Client Upgrade functionality requires performing a number of tasks using Microsoft Active Directory. For information on performing those tasks, refer to the following Microsoft Reference articles:

How To Use Group Policy to Remotely Install Software in Windows 2003.
— Article # 816102 (for Windows 2003)

ShoreTel recommends selecting the **Prevent Users from Initiating Client Upgrades** check box in the Edit System Parameters window. For details, please refer to the "Other Parameters" section of the ***ShoreTel Administration Guide***.

18.4.2 Standard Integrated Software Distribution Overview

ShoreTel system's integrated software distribution feature simplifies installation. Although the process presents a number of screens, there is a default installation that requires no input; you click through the screens until you are prompted to restart your desktop.

Users receive an email message from the ShoreTel system containing the information they need to install the ShoreTel Communicator application. The installation program is accessed using the URL listed in the email notification. Notice that the email notification includes the server name and the user name: Users will need this information later when they start the ShoreTel Communicator application for the first time.

The software can also be installed from the ShoreTel Communicator CD.

18.4.3 Installing the ShoreTel Communicator Software

You must first install the ShoreTel Communicator software.

To perform the installation:

Step 1 Go to your browser to initiate the ShoreWare client installation. Click the URL listed in your email notification, or paste (or otherwise enter) it into your web browser program (Figure 18-2).

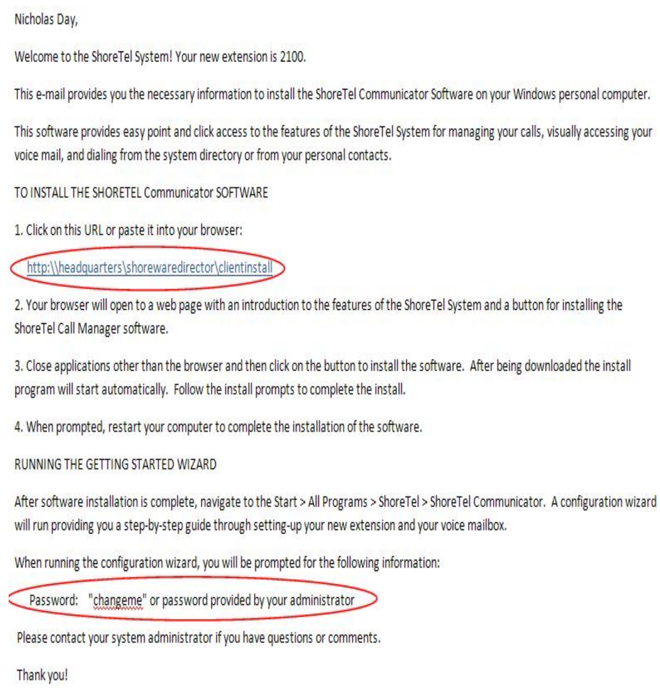
Alternatively, you can open a browser window and enter the URL http://<ShoreTel_server_name>/shoreware resources/clientinstall.

Step 2 The ShoreWare ***Client Install*** page appears. After reviewing the information on this page, click the ***Install button*** (Figure 18-3).

The InstallShield Wizard downloads the installation files (showing the progress of the download), "unpacks" the installation files, and configures the Windows Installer.

Step 3 The Welcome screen for the InstallShield Wizard appears (Figure 18-4). Notice that the version number of the ShoreWare software is shown at the bottom of the screen. To proceed, click ***Next***.

Step 4 The ShoreWare End User License Agreement appears (Figure 18-5). If you agree to the license terms, select the option **I accept the terms in the license agreement** and click **Next**.



Nicholas Day,

Welcome to the ShoreTel System! Your new extension is 2100.

This e-mail provides you the necessary information to install the ShoreTel Communicator Software on your Windows personal computer.

This software provides easy point and click access to the features of the ShoreTel System for managing your calls, visually accessing your voice mail, and dialing from the system directory or from your personal contacts.

TO INSTALL THE SHORETEL Communicator SOFTWARE

1. Click on this URL or paste it into your browser:

http:\\headquarters\shorewaredirector\clientinstall

2. Your browser will open to a web page with an introduction to the features of the ShoreTel System and a button for installing the ShoreTel Call Manager software.

3. Close applications other than the browser and then click on the button to install the software. After being downloaded the install program will start automatically. Follow the install prompts to complete the install.

4. When prompted, restart your computer to complete the installation of the software.

RUNNING THE GETTING STARTED WIZARD

After software installation is complete, navigate to the Start > All Programs > ShoreTel > ShoreTel Communicator. A configuration wizard will run providing you a step-by-step guide through setting-up your new extension and your voice mailbox.

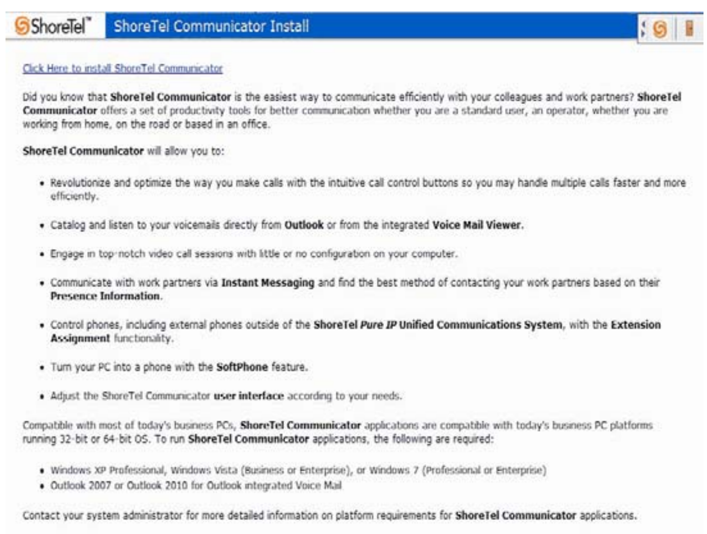
When running the configuration wizard, you will be prompted for the following information:

Password: "changeme" or password provided by your administrator

Please contact your system administrator if you have questions or comments.

Thank you!

Figure 18-2 Notification Email



ShoreTel Communicator Install

Click Here to install ShoreTel Communicator

Did you know that **ShoreTel Communicator** is the easiest way to communicate efficiently with your colleagues and work partners? **ShoreTel Communicator** offers a set of productivity tools for better communication whether you are a standard user, an operator, whether you are working from home, on the road or based in an office.

ShoreTel Communicator will allow you to:

- Revolutionize and optimize the way you make calls with the intuitive call control buttons so you may handle multiple calls faster and more efficiently.
- Catalog and listen to your voicemails directly from **Outlook** or from the integrated **Voice Mail Viewer**.
- Engage in top-notch video call sessions with little or no configuration on your computer.
- Communicate with work partners via **Instant Messaging** and find the best method of contacting your work partners based on their **Presence Information**.
- Control phones, including external phones outside of the **ShoreTel *Pure IP* Unified Communications System**, with the **Extension Assignment** functionality.
- Turn your PC into a phone with the **SoftPhone** feature.
- Adjust the ShoreTel Communicator **user interface** according to your needs.

Compatible with most of today's business PCs, **ShoreTel Communicator** applications are compatible with today's business PC platforms running 32-bit or 64-bit OS. To run **ShoreTel Communicator** applications, the following are required:

- Windows XP Professional, Windows Vista (Business or Enterprise), or Windows 7 (Professional or Enterprise)
- Outlook 2007 or Outlook 2010 for Outlook integrated Voice Mail

Contact your system administrator for more detailed information on platform requirements for **ShoreTel Communicator** applications.

Figure 18-3 Client Install page

Step 5 The InstallShield Wizard presents a default destination folder (Figure 18-6) for the ShoreWare application. Click **Change** if you want to place the ShoreTel Communicator application software in a different location. Click ***Next*** to continue.

Step 6 The ***Ready to Install*** screen appears (Figure 18-7). InstallShield has gathered enough information about your system to proceed. Click **Install** to continue.

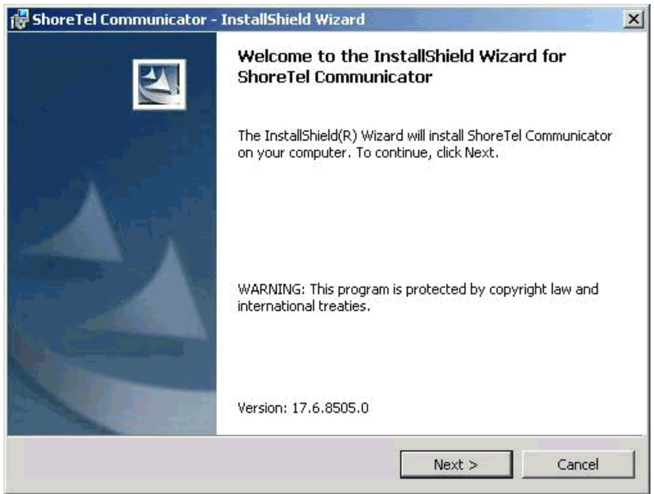


Figure 18-4 Welcome from InstallShield Wizard for ShoreTel Communicator

Figure 18-5 ShoreWare Software License Agreement

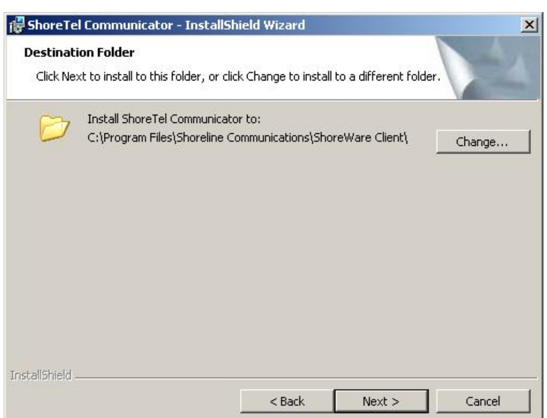


Figure 18-6 InstallShield Wizard Destination Folder

Step 7 During the final installation process, a status screen appears as shown in Figure 18-8. Installation may take a few minutes. When it is complete, click ***Next.***

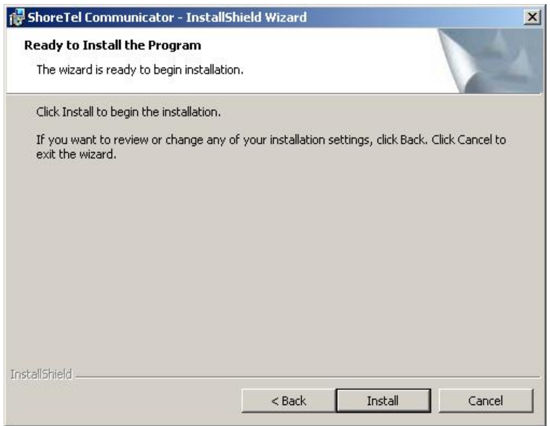


Figure 18-7 Ready to Install

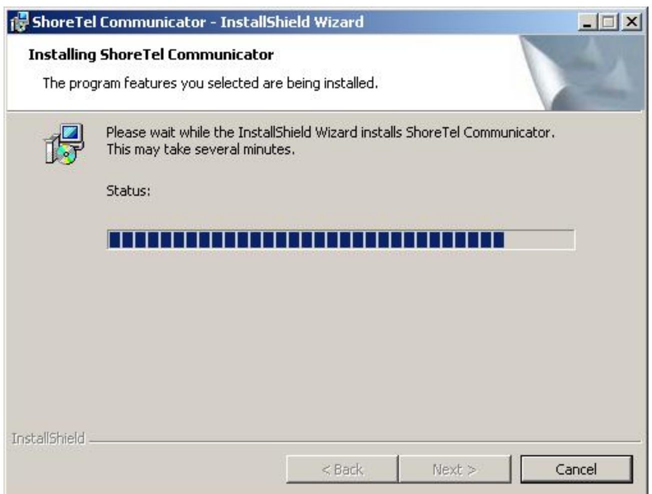


Figure 18-8 InstallShield Wizard Installation Status

Step 8 Software installation is complete when the *InstallShield Wizard Completed* screen appears (Figure 18-9). Click **Finish**.

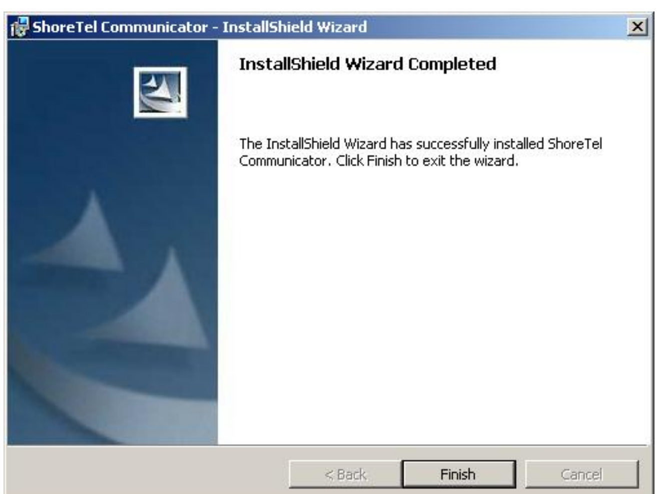


Figure 18-9 InstallShield Wizard Installation Completed

Step 9 When prompted to restart your computer (Figure 18-10), click **Yes**. The InstallShield Wizard shuts down your computer, and restarts it.

Figure 18-10 InstallShield Wizard Restart Prompt

When your desktop reappears, you will notice a new shortcut icon called ***Shortcut to ShoreTel Communicator***.

18.4.4 Configure the TAPI Dialing Parameters

The installation of the ShoreTel Communicator application will require the user to provide his area code and dialing rules if not previously configured. When this is required, the Phone and Modem Options control panel applet will start during the installation to prompt for the necessary configuration information. To continue, specify the location and area code information. Additionally, configure the dialing rules section with the appropriate information for dialing external and long distance numbers. When the information is configured and the OK button is pressed, the installation will continue.

18.4.5 Starting the ShoreTel Communicator Application

The ShoreTel Communicator application can be started in one of three ways:

- Automatically upon system startup
- From the ***Shortcut to ShoreTel Communicator*** icon on the desktop
- From the **Start > Programs > ShoreTel** menu item

The first time the ShoreTel Communicator application is started, a wizard appears, prompting you to configure your ShoreTel Communicator server, voice mail box

If you have Microsoft Outlook installed on your computer, ShoreTel Communicator will offer to install Outlook integrated voice mail. Click **Yes** to have your voice mail delivered to your Microsoft Outlook Inbox. You will also be prompted to configure AutoStart.

At this point you have completed the most typical steps associated with installing the ShoreTel Communicator application. Additional procedures are described in the following sections.

18.5 Installing Outlook Integration

You can integrate Outlook to ShoreTel Communicator in three areas: voice mail, call handling, and memorized phone number management. You can install these integrated components from the ***Outlook*** tab of the ***ShoreTel System*** dialog box.

NOTE: Users who are not local administrators will not be able to install Communicator and Outlook voice Mail/Calendar Integration.

18.5.1 Installing Voice Mail Integration

After you have installed voice mail integration, you have the option to:

- Use Outlook as the default voice mail client
- Attach voice mail to messages when moved
- Delete voice mail from messages when moved

18.5.1.1 Attach Voice Mail to Message when Moved

Check this option for your voice mail message to be saved in your Outlook folders for archival purposes. If you move a message to an Outlook folder when this option is in effect (and the Delete Voice Mail from Message when Moved option, described below, is not selected), a copy of the message is still stored on the voice mail server. If you delete the message in the voice mail interface, the Outlook copy is still available.

If you move a message without this option in effect and delete the message in the voice mail interface, the message information is still in Outlook, but the message itself is unavailable.

18.5.1.2 Delete Voice Mail from Message when Moved

Check this option to delete your voice mail messages from the ShoreTel System if you move a voice mail message to an Outlook folder. This is used to store messages in Outlook and free your voice mailbox for more messages.

To install voice mail integration:

Step 1 In the ShoreTel Communicator tool bar, click the ***ShoreTel icon***. A shortcut menu appears.

Step 2 Click ***Configure ShoreTel System***. The ***ShoreTel System*** dialog box appears.

Step 3 Click the ***Outlook*** tab as shown in Figure 18-11.

Step 4 Click ***Install***. In some cases, a warning appears requesting that you close running applications before continuing. Close the applications as requested.

18.5.2 Installing Automatic Call Handling

Although the ShoreTel Communicator installation installs the components for Microsoft Outlook integrated voice messaging, it does not install the components for the Microsoft Outlook Automated Call Handing feature. You install these components from the ShoreTel System control panel.

To install Automatic Call Handling:

Step 1 Right-click the ***ShoreTel Communicator*** icon in the Windows taskbar tray.

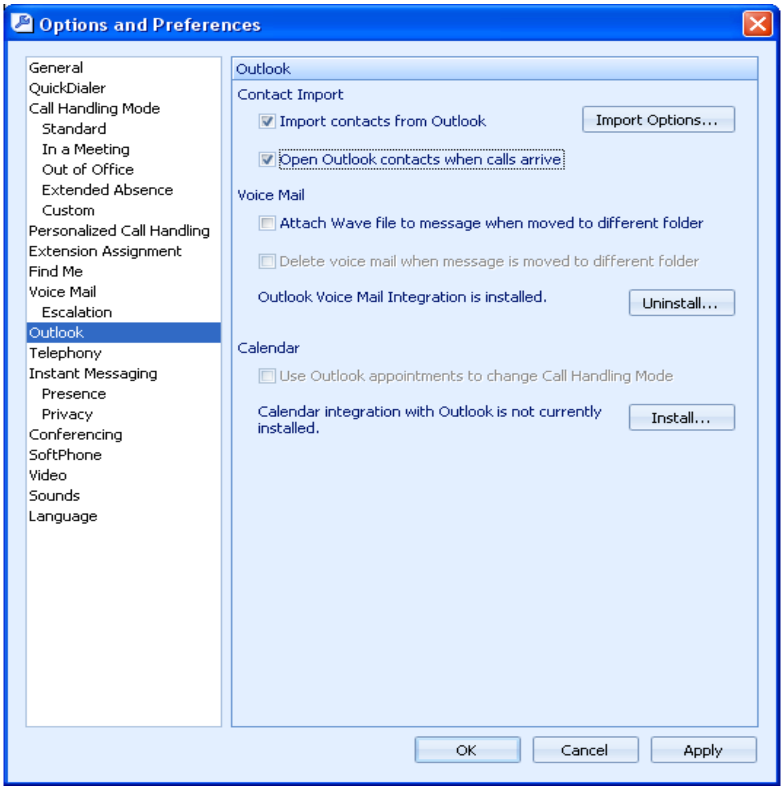


Figure 18-11 ShoreTel System Control Panel (Outlook Tab)

The ShoreTelmenu appears.

Step 2 Click *Configure ShoreTel System*.

The *ShoreTel System* dialog box appears.

Step 3 In the *ShoreTel System* dialog box, click the *Outlook* tab.

Step 4 In the *Call Handling* field, click *Install* to install the Microsoft components.

In some cases, a warning appears requesting that you close running applications before continuing. Close the applications as requested.[1]

The installation takes a few minutes to complete. Once started, it cannot be interrupted.

Collaborative Data Objects or “CDO,” a component of Microsoft Outlook, must be installed to use the automatic call handling feature. Refer to documentation on Microsoft Outlook for information on adding this component to your installation.

1. When the AutoCHM form is updated from one ShoreTel Communicator release to another, it must be re-registered on a per-user basis. This registration cannot be done by the installer. It must be done when each user logs onto ShoreTel Communicator for the first time following an upgrade. The registration requires that Outlook be closed. Users can expect to see a dialog box advising them to close Outlook if it is running at the time the registration is performed.

18.5.3 Memorized Phone Number Management

You have the option of importing Outlook contacts to the ShoreTel Communicator Quick Dial feature.

To set the option under Memorized Phone Number Management:

Step 1 In the ***Memorized Phone Number Management*** section, click ***Read phone numbers from Outlook on startup*** option.

Step 2 If you want to exclude FAX numbers from the search, click ***Don't include FAX numbers***.

Step 3 If you want Outlook Contact to appear when you have an incoming call, click ***Pop Outlook contacts on incoming call***.

Step 4 Click ***More Options*** to select which Outlook contacts to import. The MAPI Import Options dialog box appears.

Step 5 Click ***Enable Disk Caching*** if you want Outlook contacts to be available without delay when ShoreTel Communicator starts. When you have enabled disk caching, you can set when ShoreTel Communicator imports contacts. If disk caching is not enabled, ShoreTel Communicator imports contacts every time it starts.

Step 6 Click the ***Import Configurator*** tab.

Step 7 Click the locations where you want ShoreTel Communicator to search for contact information.

To select individual folders, click ***Details*** and check the folders you want searched for contact information.

Step 8 Click *OK*.

Step 9 If you want to import contacts now, return to the ***Disk Cache Options*** tab and click *Read Contacts Now*.

If you do not click this button, the Outlook contacts will be imported the next time you start ShoreTel Communicator.

It will take some time for the ShoreWare Personal ShoreTel Communicator to load your Microsoft Outlook Contacts. Your Outlook Contacts will not be available until loading has been completed.

18.6 Upgrade Procedures

When the ShoreTel system is upgraded, users running any version of ShoreTel Communicator greater than 5.5.600.0 will be informed that they must upgrade.Upgrades of the system may not require client upgrades. Refer to the online knowledge base on the ShoreCare web site to determine if a system upgrade requires client modifications.

18.7 User Licensing

ShoreTel offers three user license types:

Extension and mailbox
Extension-only
Mailbox-only

These new choices allow users to request a phone extension license without having to purchase a mailbox at the same time. This additional flexibility may be helpful in situations where a fax machine, a modem, or a lobby phone is desired and a mailbox for voice mail was not needed. Similarly, users can purchase a mailbox without having to purchase a phone extension.

Earlier releases of the ShoreTel product offered Single Site and Multi-Site Enterprise license keys. In this release, the Single Site key is no longer available. For existing users, the Single Site key can still be used and will be renamed as a “Single Site Extension and Mailbox” license. Previous Multi-Site Enterprise keys become “Extension and Mailbox” licenses.

18.7.1 Purchasing User Licenses

Each user must be configured with one of those three license types. A license must be purchased for each user, based upon the needs of that user. To see if an installation is in compliance with the number of licenses purchased, all Extension-Only, Mailbox-Only, and Mailbox-and-Extension users are counted and compared against the sum of the licenses purchased.

Extension and mailbox: Purchase of this license entitles the user to be assigned to both a physical extension and a ShoreTel mailbox.

Extension-only: Purchase of this license entitles the user to be assigned to a physical extension, either via explicit assignment or via Extension Assignment.

Mailbox-only: Purchase of this license allows the user to be assigned to a ShoreTel voice mail-box.

18.7.2 Language Licenses

ShoreTel supports Spanish, UK English, French, and German languages in addition to US English (which will remain the default language for new installations). One or more languages can be running at a site by purchasing a language license.

If only one language is needed at a single site, there is no need to purchase a language license. If Spanish or German is selected, the default language (English) must be disabled.

For instructions on configuring the User Licenses or Language Licenses via Director, please refer to the ***ShoreTel Administration Guide***.

18.7.3 License Control

License Control adds enforcement and branding to the ShoreTel product and provides tighter enforcement (via MAC address-based node locking) on existing licensing. When an existing ShoreTel system is upgraded to the current software release, an enforcement scheme requires entry of a system key.

When launching ShoreWare Director, you are asked to enter either a Small Business Edition (SBE) or Enterprise Edition (EE) key (see below for details on the differences between these two). You can request a key online via Director. If an invalid key is entered or if the field is left empty, you will be allowed to log into the system but an expiration time bomb will be activated, and you will be nagged to comply with the license requirements. If no action is taken within the 45-day grace period, ShoreWare Director will be locked and you will be unable to make any configuration changes to the system (although the phones will continue to work).

This 45-day period allows for unplanned, ad hoc changes that may cause you to exceed license limits while providing time to comply with the license requirements by either removing unneeded configurations or by ordering additional licenses.

You will be forced to purchase one of two keys available:

SBE key – required for Small Business Edition

- This key is for smaller sites that do not have remote offices
- Use of this key will result in the display of SBE branding (on the initial login page above the navigation pane)
- The number of users will be restricted to no more than 50
- Users will be unable to add an additional site key

EE key – required for Enterprise Edition

- This is for larger sites with more than one site
- Existing branding will be displayed
- System behaves as it does today, except that number of sites is enforced via nagging
- Block adding an additional SBE or EE key

Details:

For an SBE system, the following features will be disabled:

- AMIS
- SMDI
- On-net Dialing
- PSTN failover

Please refer to the ***ShoreTel Administration Guide*** for instructions on configuring SBE licensing via Director or for more information about the following types of Keyed License Types and Self-Audited License Types:

Keyed License Types:

- System License
- Additional Site License
- Extension License
- Mailbox License
- Additional Language License
- ShoreTel Communicator with Mobile Access
- ShoreTel Communicator with Professional Access
- ShoreWare SIP Phone License
- ShoreWare High Resolution Video License
- ShoreWare SIP Trunk License
- ShoreTel Communicator with Personal Access
- External Unified Messaging SIP Link license
- ShoreTel Communicator with Operator Access
- ShoreTel Communicator with WorkGroup Agent Access
- ShoreTel Communicator with Workgroup Supervisor Access

Self-Audited License Types:

— ShoreTel Communicator with Personal Access
— Remote Server Software License
— SIP Trunk License
— TAPI Application Server License
— Phone API License

18.8 Other Considerations

18.8.1 Windows Accounts and the ShoreTel Communicator

You must log in to your computer with your Windows account information to gain access to the ShoreTel Communicator application. If multiple users share the same computer, they must have separate Windows accounts to gain access to the ShoreTel Communicator application.

Be sure to install ShoreTel Communicator on the computer using the Adminaccount.When new users log in to Windows, they will see the ShoreTel Communicator icon on the desktop. The first time this ShoreTel Communicator is selected, the user is stepped through a “Getting Started” wizard.

18.8.2 Changing the Server Name

If the ShoreTel server name has changed, update the name of the server under Settings/ShoreTel login.

CHAPTER 19

Cut-Over

This chapter provides the requirements and other information for implementing the cut-over from your existing telephone system to the ShoreTel system.

19.1 Checklist

You must complete the following tasks before proceeding to the next chapter:

Task	Owner	Status
Confirm your telephony service orders with the telephone company.		
Ensure that all end-user reference guides are distributed.		
Make a copy of the site’s floor plan.		
Schedule your cut-over support.		
Test all telephones and telephone lines.		
Test the call flow, auto-attendant, and other services.		
Confirm that cut-over coverage has been assigned and scheduled.		

Table 19-1 Cutover Checklist

19.2 Cut-Over Requirements

As cut-over approaches, you should review and confirm your plan, assemble the cut-over tools, and line up resources to support the cut-over.

19.2.1 Cut-Over Worksheet

The cut-over worksheet is used by the installer during the cut-over to move all end-users from the old system to the new. It is extremely important that the cut-over worksheet be prepared before the cut-over begins. You can use the cut-over worksheet at the end of this chapter to document all new and existing connections. A soft copy of this form is available in a planning and installation workbook from ShoreTel. Make copies as necessary.

Use a pencil when preparing the cut-over worksheets, to allow for changes that may occur during the cut-over.

19.2.2 New Trunks

New trunks should be installed before cut-over. This allows time for them to be terminated, configured, and tested with the ShoreTel system.

19.2.3 Cut-Over Coverage

There are two aspects to cut-over coverage:

The team involved with planning the ShoreTel system must be on site before, during, and after cut-over.

Appropriate coverage must be scheduled to monitor the newly installed ShoreTel system for errors and last-minute configuration changes, and to help end-users with any questions they might have. ShoreTel recommends that you have support personnel on site before the first users arrive, to ensure that the system is functional and that telephone calls are processed properly.

19.3 Cut-Over Implementation

Once planning is completed, it is time to bring the ShoreTel system into service. Use the checklists in this section to implement the cut-over, starting with the top-level checklist below.

Description	Completed
Complete the tasks listed on the basic cut-over checklist.	
Cut-over and test all trunks.	
Cut-over and test the remaining devices (telephone, fax machines, modems, and so on).	
Confirm the cut-over coverage.	

Table 19-2 Cutover Implementation Checklist

19.3.1 Basic Cut-Over Checklist

Description	Completed
Secure the telephone company’s contact names, telephone numbers, and pager numbers for testing.	
Set up a command center to support cut-over activities.	
Ensure that copies of the floor plans and cut-over worksheets are available.	
Secure access to building and office areas that require ShoreGear voice switch telephones.	
Ensure that a telephone is installed next to the ShoreGear voice switch for testing.	
Ensure that music-on-hold is installed and tested.	
Record and test the auto-attendant greeting for on-hours and off-hours.	
Test all telephones.	
Test paging and night bell features, if applicable.	

Table 19-3 Basic Cut-Over Checklist

19.3.2 Trunking Cut-Over

For existing trunking, use the cut-over worksheets to identify the trunks that are used from the old system (if applicable), and terminate them on the voice switches. Use a test telephone to dial in and out of each trunk, verify that it routes to the correct location, and listen closely to the voice quality.

When preparing new trunks for installation, use the following checklist.

When all of the trunks have been tested, have the telephone company’s tester open the trunk group, and allow the callers to use the new trunks.

Description	Completed
Identify the new trunks.	
Terminate the new trunks on the ShoreGear voice switches.	
Contact the telephone company’s tester, and test each trunk (one at a time).	
Agree on the specific trunk that is being tested.	
Have the tester dial in on the new trunk.	
Answer the incoming call on a test telephone.	
Observe overall voice quality.	
Go through this checklist until all trunks are tested.	

Table 19-4 Trunking Cut-Over Checklist

19.3.3 Cut-Over of Remaining Devices

Use the following checklist to test each new end-user device that is being installed.

Description	Completed
Place an internal call from the new device.	
Place an external call from the new device.	
If applicable, place a DID call.	
If the device is for a user with voice mail, leave a welcome message similar to the following: “This is <your_name> from <company_ name>. Welcome to your new, revolutionary, IP-based communications system. You will find the following materials on your desk...”	
Leave a user guide on the user’s desk. This provides information about the ShoreTel system’s commonly used features as well as general system information.	

Table 19-5 Remaining Devices Cut-over List

19.3.4 Cut-Over Coverage

It is recommended that the cut-over team arrive on site before the beginning of the next business day after cut-over, to answer questions from end-users as they begin to use the ShoreTel system.

19.4 Cut-Over Worksheet

Name	Extension/DID	ShoreGear Port #	Patch Panel #	IDF #	Station Cable #

Name	Extension/DID	ShoreGear Port #	Patch Panel #	IDF #	Station Cable #

CHAPTER 20

Training

ShoreCare QuickStart is a virtual training program that is revolutionizing the way people learn to operate the ShoreTel system. QuickStart is an innovative, no-hassle approach to preparing system administrators, operators, and users for their ShoreTel implementation.

ShoreTel is committed to ensuring that our customers have the tools and knowledge base they need to take full advantage of the new era of communication convergence. ShoreCare QuickStart fulfills that commitment.

All the courses available through ShoreCare QuickStart are provided online for your convenience. Some instruction modules include simple interactive tutorials that introduce you to basic features and configurations of your new ShoreTel system. More advanced technical training is available via live interactive web-based sessions. In these advanced sessions you can learn about software configuration options and troubleshooting tips from an instructor providing valuable feedback for your specific issues.

For more information, please contact your ShoreTel-authorized partner or visit the ShoreCare QuickStart web center, available through www.goShoreTel.com.

20.1 Checklist

Review the following topics related to training for ShoreTel:

Task	Description
Recommendations	page 259
Training Materials	page 260
End-User Training	page 260
Operator Training	page 260
Workgroup Training	page 261
System Administrator Training	page 261

Table 20-1 Training Checklist

20.2 Recommendations

The following recommendations will assist you with training.

It is critical that all employees, workgroup agents/supervisors, and operators be familiar with ShoreTel services before the system is put in service.

Be sure to consider training needs as your staff changes over time. You can return to ShoreCare QuickStart to train new employees on the use of the ShoreTel system.

20.3 Training Materials

The following training materials are available:

User guides and self-paced online tutorials are available through the ShoreTel Communicator ***Help*** menu or from ShoreTel's online knowledge base.

System administration training and end-user training are available through a ShoreTel-authorized partner or through **ShoreTel, Inc.**

Additional training materials can be downloaded from ShoreTel.

20.4 End-User Training

QuickStart offers online tutorials to familiarize end-users with the features and functionality of the ShoreTel Communicator client. The tutorials, which are self-paced and do not require registration, highlight the commonly used features and functions available in the ShoreTel Communicator - Personal, Professional, Workgroup Agent and Operator Access. Users will learn how to install the client, answer calls, transfer a call, make conference calls, and access voice mail. A sound card and speakers are helpful but not necessary.

User training should be completed before your cut-over date.

20.5 Operator Training

Operators, receptionists, and administrative assistants have special needs and responsibilities. In addition to the ShoreTel Communicator - Operator Access tutorial, ShoreTel offers an interactive online session in which such users can learn how to maximize the power of the ShoreTel system.

ShoreTel encourages company operators, receptionists, or administrative personnel who support multiple managers to participate in a one-hour, live interactive web session introducing the ShoreTel Communicator - Operator Access . The training covers these topics:

Answering, transferring, and conferencing calls
Accessing voice mail
Using toolbar shortcuts
Monitoring extensions
Call routing
Call handling modes

Class participants are able to experience a live ShoreTel system and ask questions of the instructor.

As a prerequisite for this class, ShoreTel asks that all class participants view the ShoreTel Communicator - Operator Access tutorial.

Operator training should be completed before your cut-over date.

20.6 Workgroup Training

Workgroups, such as those in a small call center, are empowered with special features and functionality. In addition to viewing the ShoreTel Communicator - Workgroup Access tutorial, you can learn more by signing up for ShoreTel's special online training sessions on this subject.

ShoreTel encourages those customers who will be using the ShoreTel Communicator - Workgroup Access to participate in a one-hour, live interactive web session introducing the ShoreTel Communicator - Workgroup Access. These sessions are available to ShoreTel customers on a request basis and concentrate on the workgroup configuration of the requesting company.

The training covers these topics:

- Answering, transferring, and conferencing calls
- Accessing voice mail
- Using toolbar shortcuts
- Monitoring agent extensions
- Monitoring calls in the queue
- Call routing and call distribution
- Call handling modes

Class participants are able to experience a live ShoreTel system and ask questions of the instructor. Contact your ShoreTel-authorized partner or visit the ShoreCare QuickStart web center for more information regarding course content and registration.

As a prerequisite for this class, ShoreTel asks that all class members view the ShoreTel Communicator - Workgroup Access tutorial.

Workgroup training should be completed before your cut-over date.

20.7 System Administrator Training

ShoreTel welcomes system administrators to review course content and register for an interactive training session on the ShoreWare Director software. This training complements the documentation available for the system and gives system administrators the opportunity to interact with a ShoreTel system expert.

ShoreTel's system administration training is designed for IT professionals who will be responsible for the configuration and ongoing support of the ShoreTel system. The training covers these topics:

- Getting started
- Setting up single-site and multisite environments
- Configuring ShoreGear switches
- Trunks
- Users
- Voice mail
- Automated attendant menus
- Workgroups
- Maintenance

The class (led by an online instructor) lasts about four hours. Participants are able to interact with a ShoreTel system and ask questions of the instructor. Contact your ShoreTel-authorized partner or visit the ShoreCare QuickStart web center for more information regarding course content and registration.

Please register for system administration training at least three weeks before your proposed cut-over date.

APPENDIX *A*

International Planning and Installation

This chapter provides detailed information about voice switches, operating systems, and features that are supported when the ShoreTel system is used outside the United States of America.

A.1 Software and Feature Support

For information concerning software and feature support throughout the world, refer to the ShoreTel 11.1 Country Availability chart located at

http://partners.shoretel.com/product_sales_tools/releases/downloads/shoretel_11.1_country_availability.pdf

A.2 Language Packs

Language packs within the ShoreTel system define the language used in the following three independent parts:

- Voice prompts (Voice mail, Auto Attendant, system announcements)
- Telephone User Interface (telephone display and ShoreTel Communicator interface)
- Online help for ShoreTel Communicator

Language pack availability affects the behavior of the system in the following areas:

- Site
- Trunk
- Workgroup
- Auto Attendant
- Voice Mail
- User
- ShoreTel Communicator

Director panels that program language options include:

Edit Site panel: The Edit Site panel, shown in Figure A-1, specifies the language pack used by the Backup Auto-Attendant (BAA).

To access the Edit Site panel, select ***Administration -> Site*** from the main menu, then click on the name of the desired site.

Workgroup panel: The Workgroup panel, shown in Figure A-2, specifies the language that the system uses for playing prompts to inbound callers.

Figure A-1 Language set at Site level

To access the Edit Workgroup panel, select *Administration -> Workgroups* from the main menu, then click on the name of the desired site.

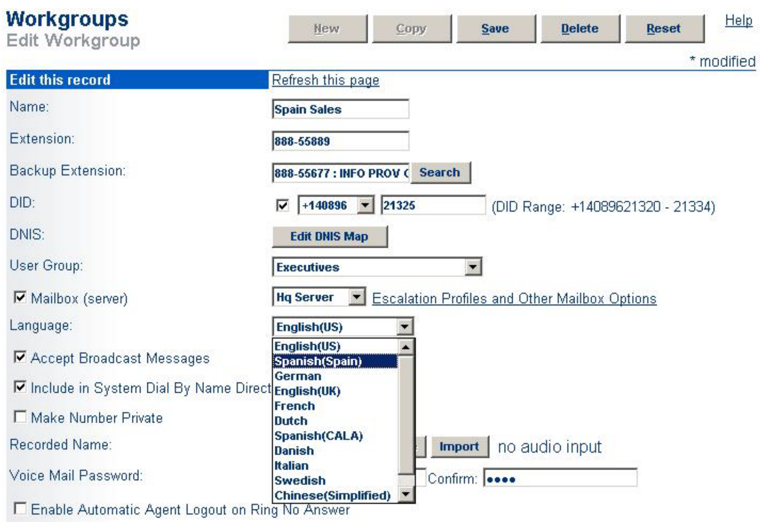


Figure A-2 Language set at Workgroup level

Edit Trunk Group panel: The Edit Trunk Group panel, shown in Figure A-3, specifies the language prompts are played to incoming callers.

To access the Edit Trunk Group panel, select *Administration -> Trunks -> Trunk Groups* from the main menu, then click on the name of the desired trunk group.

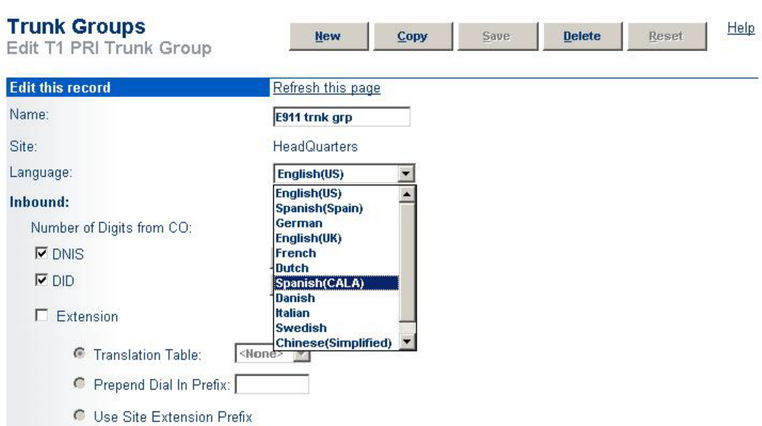


Figure A-3 Language set at Trunk Group level

Edit User panel: The Edit User panel, shown in Figure A-4, specifies the language prompts used for the user's telephone interface and voicemail prompts.

To access the Edit User panel, select *Administration -> Users -> Individual Users ->* from the main menu, then click on the name of the desired user.

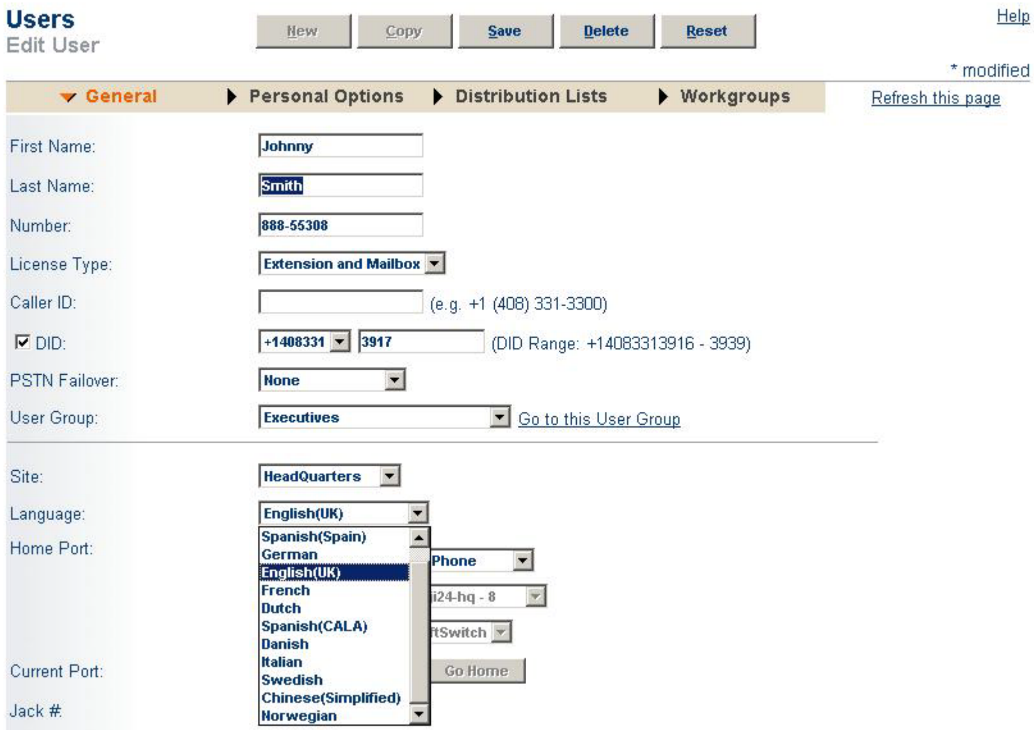


Figure A-4 Language set at User level

In language priority, a workgroup language overrides the language associated with a trunk, which in turn overrides the language associated with an individual user.

A.3 Analog Telephones, Tones, Cadences, and Impedances

For all supported countries, standard analog telephones are available on a per-country basis. The main difference between telephones in different countries is the line impedance. The ShoreWare Distributed Call Control software will provide the appropriate impedance required for each supported country. Tones, cadences, and impedance requirements are matched on a per-country basis.

A.4 Dialing Plan Considerations

When planning a global voice network, remember that the ShoreTel system is a single image system and that you must consider all countries and locations when designing the international dialing plan. The ShoreTel system can match the dialing plan requirements of the local service provider for the supported countries.

A.4.1 Single-Extension Plan

Across the global voice network, all extensions must be unique and cannot overlap.

A.4.2 Trunk Access Codes

Across the global voice network, when you configure trunk access codes, that portion of the dialing plan will be reserved so you will be sacrificing one digit. Typically in the US, customers use 9 as a trunk access code. Internationally, those in the EMEA, for instance, often use 0 as a trunk access code

Using two different trunk access codes will limit users to only being able to access certain trunk groups.

If you use a single trunk access code, some users will need to be retrained.

Alternatively, 8 could be defined for the trunk access code globally.

ShoreTel recommends proper identification **from the beginning**. The trunk access code should not be changed later.

A.4.3 Operator Digit

The leading digit of 0 is typically reserved for dialing the operator in the US. The operator digit is configurable. Similarly, EMEA customers are accustomed to dialing 9 to reach the operator.

ShoreTel recommends choosing a single digit for the trunk access code and selecting a different single digit for the operator.

A.4.4 Emergency Numbers

The ShoreTel system allows dialing of emergency numbers with and without trunk access codes. For this reason, you should architect the dialing plan for this feature.

911 is used in the US.
112 is used in Europe and other countries.
Check for Asia per local requirements.

Thus, extensions should not begin with 0, 1, or 9 to make use of this feature.

Each site can have a maximum of ten emergency numbers to accommodate locations where multiple emergency service numbers are required.

For more information about emergency numbers, see “Emergency 911 Operations” appendix in the ***ShoreTel Administration Guide***.

A.4.5 DID Numbers

DID numbers are related to the trunk group in which they are associated. You should strive to match the last digits of the DID number to the user’s extension number.

APPENDIX B

Regulatory and Safety Information

This chapter provides detailed information regarding compliance of the ShoreTel system with the international regulatory bodies. The chapter also addresses safety concerns related to installation, operation and general use of the ShoreTel system.

B.1 Agency Approvals

Category	Regulatory Compliance / Agency Approval
EMC	EN 55022 Class A (SG-12, SG-8, SG-24, SG-T1)
	EN 55022 Class B/Class A (SG-E1)
	FCC Part 15 Class A (SG-12, SG-8, SG-24, SG-T1)
	EN 55024:1998 including amendments A1:2001 and A2:2003 (SG-24, SG-E1)
Electrical Safety	FCC Part 68 for SG-24, SG-T1
	IEC 60950:1999 3rd ed. SG-8, SG-12, SG-24, SG-T1, SG-E1
	EN60950:2000 SG-8, SG-24, SG-T1, SG-E1
	AS/NZ 60950:2000 SG-8, SG-24, SG-T1, SG-E1
	UL60950 3rd ed. 2000 SG-8, SG-12, SG-24, SG-T1, SG-E1
	ACA TS001-1997: SG-8, SG-24, SG-T1, SG-E1
	FCC Part 68: SG-8, SG-12, SG-24
Telecom	ETSI TS 103 021-1 V1.1 (2003-08) SG-8, SG-12, SG-24
	ETSI TS 103 021-2 V1.2 (2003-09) SG-8, SG-12, SG-24

Table B-1 Agency approvals

Category	Regulatory Compliance / Agency Approval
	ETSI TS 103 021-3 V1.2 (2003-09) SG-8, SG-12, SG-24
	ETSI TBR4 Nov. 1995 SG-E1
	ETSI TBR4/A1 Dec. 1997 SG-E1
	ETSI TS 102 119 V.1.1.1 Aug. 2001 SG-E1
	Bellcore GR-499-CORE, issue 2, Dec. 1998 SG-T1
Telecom Homologation	NZ PTC 220/06/016 through PTC 220/06/023
	SG-8, SG-12, SG-24, SG-T1, SG-E1 and IP 210/530/560 Phones

Table B-1 Agency approvals

B.2 EMC Compliance Statements (SG-8/12/24 and T1)

B.2.1 United States

This equipment has been tested and found to comply with the limits for Class A digital devices, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

Changes or modifications to this equipment not expressly approved by the party responsible for compliance to FCC part 15 could void the user's authority to operate the equipment.

B.2.2 European Union

This is a Class A product. In a domestic environment this product may cause radio interference in which case the user may be required to take adequate measures.

B.2.3 Canada

This Class A digital apparatus complies with Canadian ICES-003. Cet appareil numérique de la classe A est conforme à la norme NMB-003 du Canada.

B.2.4 Restricted Access Location

This product is intended to be installed only in a RESTRICTED ACCESS LOCATION. A RESTRICTED ACCESS LOCATION is defined as an area where access can be gained only by SERVICE PERSONNEL who have been instructed about the reasons for the restrictions applied to the location and about any precautions that must be taken. RESTRICTED ACCESS LOCATIONS can be accessed only through the use of a tool or lock and key or other means of security, and are controlled by the authority responsible for the location. SERVICE PERSONNEL are defined as persons having appropriate technical training and experience necessary to be aware of hazards to which they are exposed in performing a task and of measures to minimize the danger to themselves or other persons.

B.2.5 WEEE Information

In accordance with the requirements of council directive 2002/96/EC on Waste of Electrical and Electronic Equipment (WEEE), ensure that at end-of-life you separate this product from other waste and scrap and deliver to the WEEE collection system in your country for recycling.

B.3 Safety

The following information is included in this publication for the use and safety of installation and maintenance personnel.

WARNING This equipment uses a three-conductor power cord with safety grounding conductor. Ensure that this is connected to an AC outlet with provision for grounding. Ensure the permanent earthing protector is connected as directed in the installation instructions. Consult a licensed electrician if necessary.

B.3.1 Important Safety Instructions

Read all of the instructions before attempting to operate the equipment and before connecting the power supply.

Always follow basic safety precautions to reduce the risk of fire, electrical shock, and injury to persons.

To prevent fire or shock hazard, do not expose the unit to rain, moisture, or install this product near water. Never spill liquid of any kind on this product.

Never push objects of any kind into this product through openings, as they may touch dangerous voltage points or short out parts, which could result in the risk of fire or electrical shock.

Do not open the cabinet, as there are high voltage components inside. Refer servicing to qualified service personnel.

Do not attach the power supply cord to building surfaces. Do not allow anything to rest on the power cord or allow the cord to be abused by persons walking on it.

To protect this equipment from overheating, do not block the openings in the housing that are provided for ventilation.

B.3.2 Electrical Safety

WARNING Do not take chances with your life. Follow these guidelines carefully:

Observe all safety regulations and read the warnings, cautions, and notes posted on the equipment.

Never assume that the power is turned off. Always check to ensure that a circuit does not have power.

Connect all power before installing changes in systems or wiring.

Use caution when installing or modifying telephone lines. Never install telephone wiring during an electrical storm.

Never touch uninsulated telephone wires or terminals unless the telephone line has been disconnected at the network interface.

Telephone connections to the unit should be made with number 26 AWG wire in order to minimize risk of fire.

APPENDIX C

IP Phone Configuration

ShoreTel IP phones are preconfigured by ShoreTel to work in conjunction with your ShoreTel system and your network's Dynamic Host Configuration Protocol (DHCP) server. Once the servers are configured, you simply plug the phones into the network and they are automatically added to your ShoreTel system.

The ShoreTel server provides the IP phones with the latest application software and the configuration information that enables the IP phone to be automatically added to the ShoreTel system. The ShoreTel server's address must be provided to the phone as a vendor-specific option.

For information on configuring DHCP for the IP phones, see Section 9.7 on page 123 and Section 16.3.6 on page 228.

However, if you are installing ShorePhone IP phones in a network without a DHCP server, you must set the IP parameters manually through the phone interface.

C.1 Manually Configuring ShorePhones

If you are not using a DHCP server to provide the IP address and configuration parameters to the phone, you need to manually set configuration parameters on the phone.

You can enter the phone configuration menu at bootup or by entering a key sequence from the phone's keypad.

To manually configure a ShorePhone at bootup:

Step 1 Connect the Ethernet cable into the data jack on the back of the IP phone or BB24 device.

Step 2 At the **Password** prompt, enter the default password **1234**, or the password provided by your system administrator, followed by the # key.

You have four seconds to enter the password, after which the phone enters normal operation with its current settings.

The default password can be changed in ShoreWare Director. For more information, see the ***ShoreTel Administration Guide***.

The BB24 setup screen can be accessed by pressing the upper leftmost and lower rightmost buttons.

Step 3 Enter the values listed in Table C-1 when prompted. Press # to advance to the next settings or * to exit.

The phone downloads the latest bootROM and firmware from the ShoreTel server and in the process, reboots several times. When the phone displays the date and time, the boot and upgrade process is complete.

Prompt	Value
Clear All Values?	Press #. (No)
DHCP=	Press * to toggle to the "off" position and then press #.
FTP=	Enter the IP address of your ShoreWare server. Press #.
MGC=	Press #. (The phone obtains the address from configuration files on the ShoreWare server).
SNTP=	Enter the IP address of your time server. Press #.
802.1Q Tagging=off	Press #. Consult your network administrator before changing this value.
VLAN ID=	Press #.
Country=	Enter the country code (see Table 9-12 on page 124).
Language=	Enter the language code (see Table 9-13 on page 125).
Save all Changes	Press #. (Yes)

Table C-1 Configuration Values

To manually configure an operational ShorePhone from the keypad:

Step 1 With the phone on hook, press the MUTE key followed by 73887# (SETUP#).

Step 2 At the **Password** prompt, enter **1234**, or the password provided by your system administrator, followed by the # key.

The default password can be changed in ShoreWare Director. For more information, see the ***ShoreTel Administration Guide***.

Step 3 Enter the values listed in Table C-1 when prompted. Press # to advance to the next settings or * to exit.

The phone downloads the latest bootROM and firmware from the ShoreTel server and in the process, reboots several times. When the phone displays the date and time, the boot and upgrade process is complete.

C.2 Displaying ShorePhone 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 With the phone on hook, 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.

C.2.1 Resetting a ShorePhone

You can reset the phone by entering a key sequence from the phone's keypad.

To reset the phone:

Step 1 With the phone on hook, press the MUTE key followed by 73738# (RESET#). The phone will reboot.

APPENDIX D

Enabling Internet Access to ShoreTel Communicator for Web

This appendix describes how to provide Internet access to ShoreTel Communicator for Web client using Apache Server as a reverse proxy.

D.1 Overview

ShoreTel recommends that you enable Internet access to ShoreTel Communicator for Web by deploying a reverse proxy based in the DMZ of your corporate firewall. You can use any of the many reverse proxy products available to implement this solution.

This appendix provides information specific to Apache Server installed on a Microsoft 2003 Server. Apache Server is an open source product and is widely used today. Additional product details and information for the Apache Server can be found on the Apache Web site.

WARNING Implementing a reverse proxy server incorrectly can compromise the security of your corporate network. Before attempting to implement a reverse proxy server, consult a network security expert with proxy and firewall experience. Open proxy servers present vulnerabilities to both the private corporate network and the public Internet.

D.2 Requirements

To complete the implementation described in this appendix you need:

Windows Server 2003. Additional OS platforms are supported.
Apache Version 2.x or later

D.3 Installation and Configuration

The following sample configuration is based on the Apache Server sitting in a DMZ with a legitimate Internet IP address.

To install an Apache Server as a reverse proxy:

Step 1 Install the Apache Server. For proper installation and setup, see Apache documentation.

Step 2 After you have installed the Apache Server, find the Apache documentation and read the Proxy Module section. The default location for the Proxy Module documentation is: `http://servername/manual/mod/mod_proxy.html`. Read the entire section before continuing.

Step 3 Open the `httpd.conf` file (see Apache documentation for location of the `httpd.conf` file).

Step 4 Add the lines from Example 1 or Example 2 to the end of the file.

Example 1 uses the default HTTP port 80. Example 2 uses port 5440, which is a port director that CSIS and ShoreWare Web Client monitor.

In the examples given below, replace the text "ServerName" with the machine name or IP address of the ShoreWare Director server.

Step 5 Depending on which port you are using, either port 80 or port 5440, you must open the firewall to allow traffic from the proxy to the ShoreWare server.

D.3.1 Example 1

```
########################################################################
#Load the general proxy module and the http specific one
LoadModule proxy_module modules/mod_proxy.so
LoadModule proxy_http_module modules/mod_proxy_http.so
########################################################################
#make sure you disable forward proxy
ProxyRequests off

#Reverse proxy to ShoreTel Web Client
ProxyPass /ShoreWareWebClient/ http://ServerName/ShoreWareWebClient/
ProxyPassReverse /ShoreWareWebClient/ http://ServerName/ShoreWareWebclient/

# Note: This configuration will use the default HTTP port 80
```

D.3.2 Example 2:

```
########################################################################
#Load the general proxy module and the http specific one
LoadModule proxy_module modules/mod_proxy.so
LoadModule proxy_http_module modules/mod_proxy_http.so
########################################################################
#make sure you disable forward proxy
ProxyRequests off

#Reverse proxy to ShoreTel Web Client
ProxyPass /ShoreWareWebClient/ http://ServerName:5440/ShoreWareWebClient/
```

```
ProxyPassReverse /ShoreWareWebClient/ http://ServerName:5440/
ShoreWareWebClient/
```

D.3.3 About the httpd.conf file

In the above examples, setting "ProxyRequests" to "off" prevents the Apache Server from functioning as a forward proxy server. This setting does not disable use of the ProxyPass directive.

In a typical reverse proxy configuration, this option should be set to "off."

If you want the additional functionality of HTTP or FTP proxy sites, add the following lines to the configuration file:

```
mod_proxy_http <../mod/mod_proxy_http.html>
```

or

```
mod_proxy_ftp <../mod/mod_proxy_ftp.html>
```

APPENDIX E

ShoreWare Clients on Citrix and Windows Terminal Servers

This appendix describes how to configure Citrix and Windows Terminal Servers to run ShoreTel Communicator clients.

E.1 Overview

Windows Terminal Server (WTS) and Citrix technologies can dramatically reduce management overhead in environments where many users use the same set of applications on similar PC desktops. These technologies allow you to centralize applications and simplify application management and upgrades. Additionally, these technologies allow you to remotely assist and support users with application questions or issues.

This appendix provides information specific to running ShoreWare clients. For complete information on Windows Terminal Server or Citrix technologies, see the documentation available online at the Microsoft or Citrix Web sites.

E.2 Citrix XenApp Environment Best Practices

ShoreTel recommends the following best practice guidelines for computers running ShoreTel on XenApp servers:

- Use only Citrix-ready anti-virus on their XenApp server
- Run XenApp and ShoreWare servers on a Citrix qualified server platform

E.3 Installing ShoreTel Communicator on WTS or Citrix

ShoreTel supports ShoreTel Communicator on the following platforms:

- WTS 32-bit
- WTS 64-bit
- Citrix 32-bit

E.3.1 Initial Installation (all platforms) and Upgrades (32-bit)

The following procedure is required when installing or upgrading ShoreTel Communicator on 32-bit platforms or when installing ShoreTel Communicator on 64-bit platforms. For instructions on upgrading ShoreTel Communicator on 64-bit platforms, refer to Section E.3.2.

Step 1 Install ShoreWare client as described in Chapter 18, starting on page 239.

Step 2 Reboot if requested.

Step 3 Go to the Windows Control Panel and open the ***Phone and Modem Options -> Advanced*** tab as shown in Figure E-1

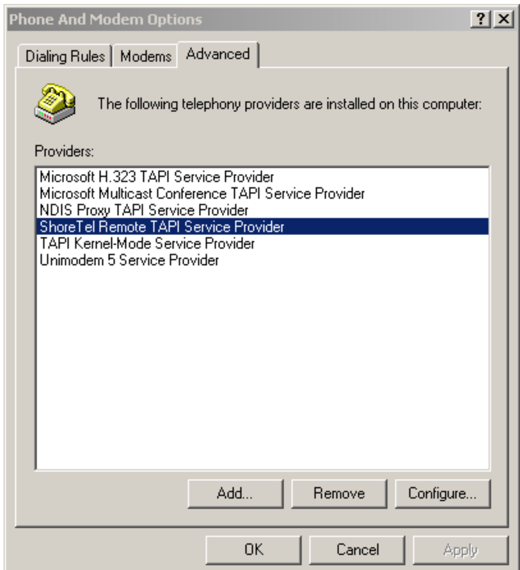


Figure E-1 Phone And Modem Options, Advanced tab

Step 4 Remove all ShoreTel providers.

Step 5 Copy the following file “TspInstall.exe” from the headquarters machine (***Program Files > Shoreline Communications > ShoreWare Server***) to the Citrix terminal server. We recommend copying the file to the following location:

```
c:\program files\Shoreline Communications\ShoreWare Client\
```

Step 6 From the Citrix terminal server, launch the command prompt by clicking on the ***Start*** bar and selecting ***Run*** and typing **cmd**.

Step 7 Navigate to the directory where the “TspInstall.exe” file was copied and run the TSPinstall utility as shown in Figure E-1. Make sure you substitute the correct hostname or IP address of the Headquarters instance of ShoreWare Server. The syntax of the command is:

```
TSPinstall -i StServer <HQ servername>
```

Step 8 Return to the Windows Control Panel and open the ***Phone and Modem Options -> Advanced*** tab.

Step 9 Click on the ***ShoreTel provider*** and click ***Configure*** to display the ***ShoreTel Remote TSP*** dialog box, as shown in Figure E-3.

Step 10 If the ShoreTel Remote TAPI Service Provider is connected to the ShoreWare Server, the ***ShoreTel Remote TSP*** dialog box appears as shown in Figure E-4.

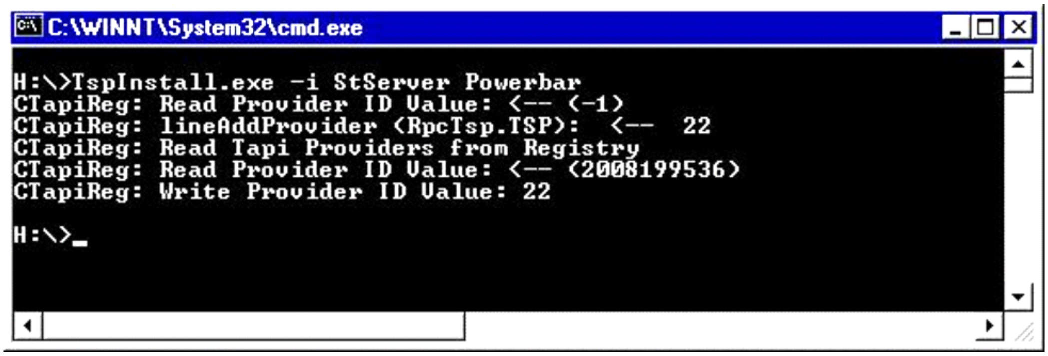


Figure E-2 TSPInstall Command Line

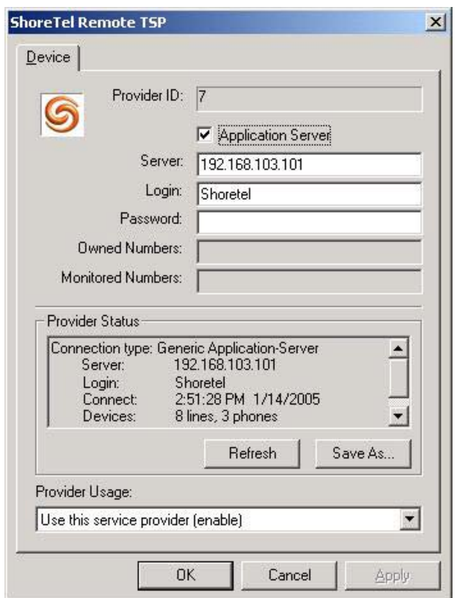


Figure E-3 Functioning Remote TSP under WTS and Citrix Example

Figure E-4 shows an error message in the ***Provider Status*** field and has blanks for the Server Name and Login fields. This indicates a null instance of ShoreTel Remote TAPI Service Provider and that this provider must be removed.

Step 11 To remove the provider, go to the ***Control Panel*** and open ***Phone and Modem Options***, and then select the ***Advanced*** tab. Click on ***ShoreTel Remote TAPI Service Provider*** and then click ***Remove***, as shown in Figure E-1.

E.3.2 Upgrading ShoreTel Communicator on 64-bit Platforms

To upgrade ShoreTel Communicator on 64-bit platforms, perform the following procedure:

Step 1 Go to the Windows Control Panel and open the ***Phone and Modem Options -> Advanced*** tab as shown in Figure E-1

Step 2 Remove all ShoreTel providers.

Step 3 Perform the procedure in Section E.3.2, starting with Step 1.

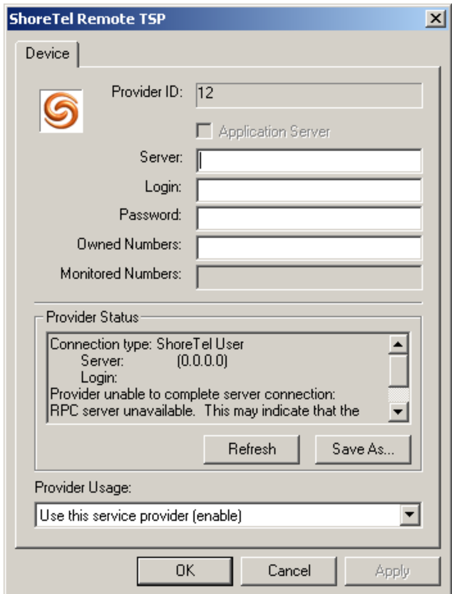


Figure E-4 Non-functioning Remote TSP for an Application Server

E.4 Citrix Application Mode

Citrix supports two application modes: Desktop and Published Application.

For more information, refer to the XenApp Product bulletin available at support.shoretel.com.

E.5 Configuring Other TAPI Applications

Unlike the ShoreTel Communicator installation on a single user system, the TAPI Service Provider on a Windows Terminal Server or Citrix system provides access to all telephony endpoints. While the ShoreTel Communicator application only accesses the telephone extension for the appropriate user, care must be taken with third-party TAPI-capable applications which may be configured to act on any telephone extension.

For example, Micrsoft Outlook and the Windows Dialer can be configured to place calls on a ShoreTel extension via TAPI. Each of these applications must be configured on a per-user basis to use the correct line device for that user. Once the Windows Dialer has been configured, it will store a unique line identifier in the Windows Registry for that user so that future sessions will always use the correct telephone extension.

APPENDIX F

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.

F.1 Overview

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.

In ShoreTel 7.5, SIP is supported via SIP trunks. 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.

F.1.1 Supported RFCs

ShoreTel supports 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

F.1.2 General SIP Comments

F.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.

F.1.2.2 DTMF

With G.729, ShoreTel both sends and receives DTMF out of band per RFC 2833.

With G.711, ShoreTel will only receive DTMF per RFC 2833. Not all ShoreTel endpoints will send DTMF with G.711. For example, switches may not but ShoreTel IP phones will.

ShoreTel IP phones support in-band G711 DTMF signaling. However, out-of-band DTMF is required for a SIP device to send DTMF to ShoreTel's voicemail or auto attendant. SIP INFO or DTMF per RFC 2833 can be used.

ShoreTel can be configured to use the SIP INFO function for DTMF signaling for environments where out-of-band DTMF is needed but in which RFC 2833 is not applicable. Note that SIP tie trunks must use SIP INFO and cannot use RFC 2833 DTMF Relay.

ShoreTel offers support for RFC2833 (DTMF), so if the voicemail server is down, external callers can now enter an extension using DTMF to ring the extension of the user they are trying to reach. This allows callers who are accessing the ShoreTel system over a SIP trunk to have access to the Backup Auto-Attendant in the same manner as users who are accessing the system via all other trunk types.

F.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.

F.1.2.4 Routing with Static and Dynamic Trunks

From the trunk group perspective, when static and dynamic trunks are used:

— only one trunk group with dynamic trunks is allowed per switch

outbound calls to this trunk group must be completed based in the registration table

calls to the same IP address will not work

calls to different devices going to the PSTN will be selected randomly

— 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

— OSE's routed over trunk groups in more than one switch (with dynamic trunks) will fail

F.1.2.5 General Feature Limitations

Incoming calls to an IP phone placed over a SIP tie trunk (via G729) require the IP phone user to press the “To VM” soft key twice in order to successfully transfer the caller to voice mail.

ShoreTel introduces support for Music On Hold (MOH) over SIP trunks. The capacity limits of MOH switches will not change (i.e. a switch will still be capable of providing up to 15 streams). However, these streams can be to other switches or to SIP devices, so customers who were not at the switch capacity limit before may now find themselves testing the limits of the switch capacity.

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.

3-way conference on a SIP trunk call uses Make Me conference ports. A minimum of 3 Make Me ports must be configured to support 3-way conferencing. Make Me conferencing for 4 to 6 parties is not supported.

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 is 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.

Park/Unpark is not supported on a SIP trunk call. This is planned for a future release.

Extension Assignment is not supported on SIP trunks. Outbound trunk hunting will automatically avoid SIP trunks when placing the call to the Extension Assignment user. The call to the Extension Assignment user cannot be a SIP trunk; however, the call to the external party can be a SIP trunk.

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.

ShoreTel SIP supports two codecs - G.711 and G.729.

G.711 SIP devices that do not support RFC 2833 DTMF cannot send DTMF digits to Voicemail (VM) or Auto-Attendants (AA).

G.729 only SIP devices cannot talk to VM/AA unless they are configured as Teleworkers or configured in remote site.

F.1.2.6 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 ShoreGear 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 ShoreGear 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 must be deleted individually (i.e. 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.

Can be found in Director under ***Call Control > Options.***

F.2 Configuration

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)

Configure the SIP Device (per the manufacturer's instructions)

These tasks will be discussed in more detail below.

F.2.1 Configuring the ShoreTel System via Director

One of the first tasks involved in configuring your ShoreTel system for Session Initiation Protocol (SIP) is to reserve the trunk. This can be done by following the procedure below:

F.2.1.1 Reserve the SIP Trunk

To reserve a new SIP trunk, follow the procedure below:

Step 1 Launch **ShoreWare 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 **Switches** link.

Step 4 Click on the **Add new switch at site** drop-down menu and select the location for the new switch, or edit an existing switch.

Step 5 Click on the **of type** drop-down menu and select the type of switch that will be used to support the SIP trunks.

Step 6 Click **Go** to display a window similar to the one shown below.

Step 7 Enter a name for the switch in the **Name** field.

Step 8 Enter a description for the switch in the **Description** field.

Step 9 Click the **Find Switches** button next to the IP Address field and select the appropriate switch to populate the field with an IP address.

Step 10 The Ethernet Address field auto-populates.

Step 11 Click on the **Server to Manage** drop-down menu and select the server that will manage this switch.

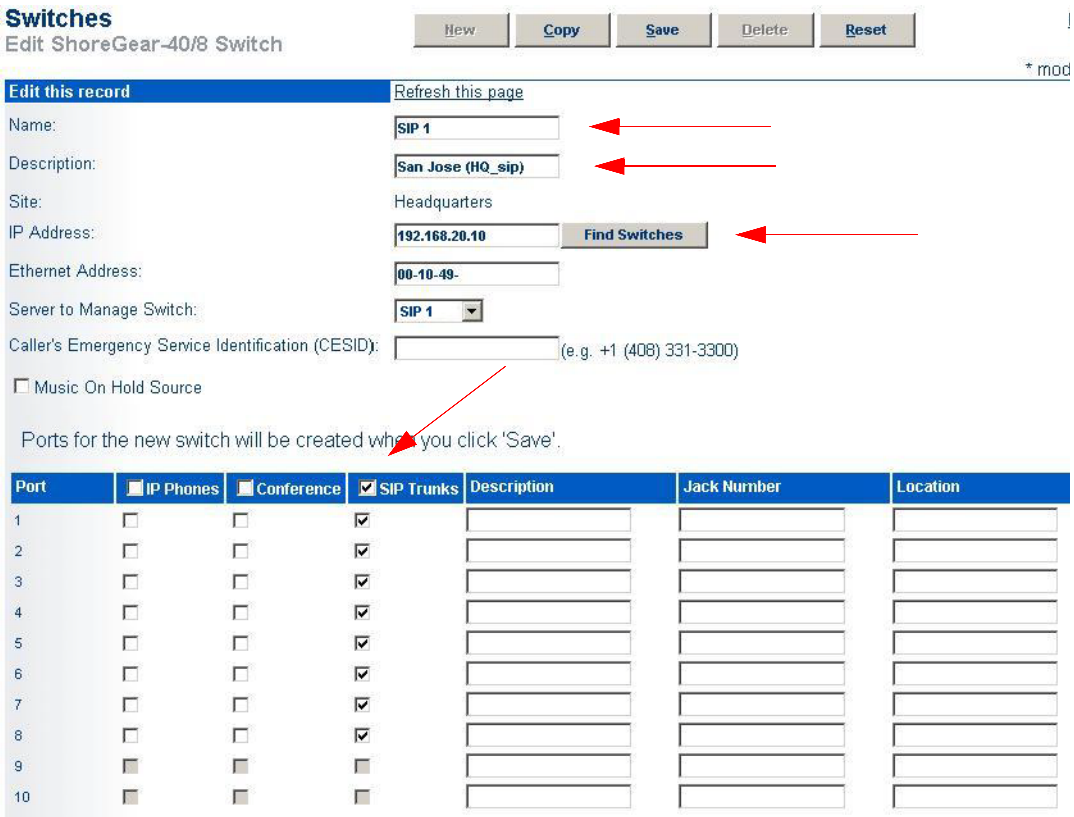


Figure F-1 Configuring switch for SIP

Step 12 Enter a CESID value in the **Caller's Emergency Service Identification** (CESID) field. (optional)

Step 13 Select the **SIP Trunks** check box to the right of the port(s) that will be associated with a SIP trunk. Each port supports five SIP trunks.

The fax redirect feature will not work with calls that come in on SIP trunks.

Step 14 Click the **Save** button to store your changes.

F.2.1.2 Create a SIP Trunk Group

To create a new SIP trunk group, follow the procedure below:

Step 1 Launch **ShoreWare 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 **Trunks** link to expand the list.

Step 4 Click on the **Trunk Groups** link.

Step 5 In the **Add new trunk group at site** drop-down menu, select the location where the new SIP trunk group will be added. In the of type drop-down menu, select SIP.

Step 6 Click **Go** to display a window similar to the one shown below.

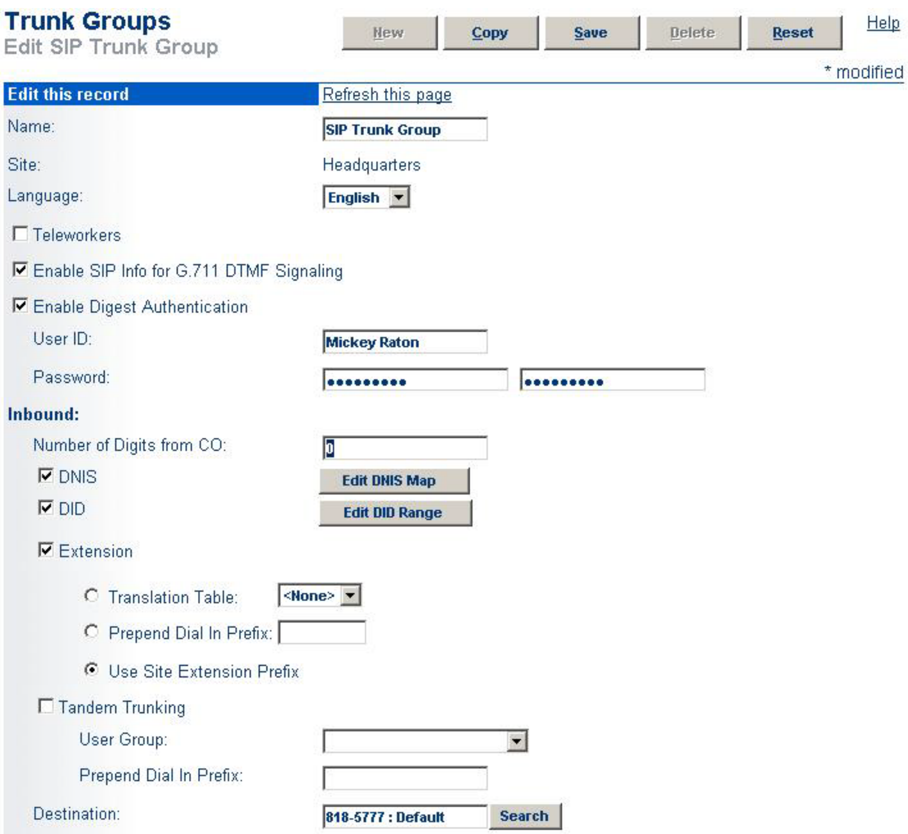


Figure F-2 **Creating a SIP trunk group (inbound configurations)**

Step 7 Enter the name of the trunk group in the **Name** field.

Step 8 Select the desired language for the trunk group in the **Language** drop-down menu.

Step 9 Select the **Teleworker** check box if the SIP endpoint is not at the same site as the trunk group being configured. Selecting **Teleworker** has the following effects on the system behavior:

Audio proxies via the SG vs. the RTP are directed to ShoreWare 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 10 Select 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 two ShoreTel systems with SIP tie trunks. Clear if the trunk is primarily used to connect a third-party SIP device.

Step 11 Select the **Enable Digest Authentication** check box and enter a user ID and password for enhanced security. All third-party SIP devices will be required to 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. (optional)

If checked, any third-party SIP devices that you would like to have access the ShoreTel system must be configured with the same user ID and password information that you have entered here.

Step 12 Enter the desired number in the **Number of Digits from CO** field.

Step 13 Select the **DNIS** check box and click the Edit DNIS Map button to add entries to the DNIS Map.

Step 14 Select the **DID** check box and click the Edit DID Range button to add entries to the DID Digit Map.

Step 15 Select the **Extension** check box to route calls directly to the extension based on the number of digits received from the SIP device, and select the appropriate radio button.

Translation Table - Select this option to use a digit translation table to ensure that inbound calls are the proper length.

Prepend Dial in Prefix - Select this to prepend inbound calls with a number that you can specify in the field.

Use Site Extension Prefix - Select this to use the extension prefix associated with the site.

Step 16 Select the **Tandem Trunking** check box allow a legacy voice system to use a ShoreTel system for outbound dialing.

User Group - Tandem calls are associated with a user group for outbound trunk selection. In-bound 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.

Dial in Prefix - When needed, you can specify a “dial in prefix” which is prepended to digits collected on tandem calls. The concatenated set of digits is then be used in outbound trunk selection for the tandem call.

Step 17 Click **Save** to store your changes.

To configure the outbound options for this trunk group:

Step 1 Continue scrolling down to display a window similar to the one below:

Step 2 Enter the appropriate trunk access code for this trunk group in the **Access Code** field. This is typically “9” in the U.S. and Canada.

Step 3 Enter the local area code for this trunk group in the **Local Area Code** field.

Step 4 Select the **Local** check box to enable local calls.

Step 5 Select the **Long Distance** check box to enable long-distance calls.

Step 6 Select the **International** check box to enable international calls.

Step 7 Select the **n11** check box to enable telephone service calls, such as directory assistance (e.g., 411 or 611, but not 911, which is specified below.)

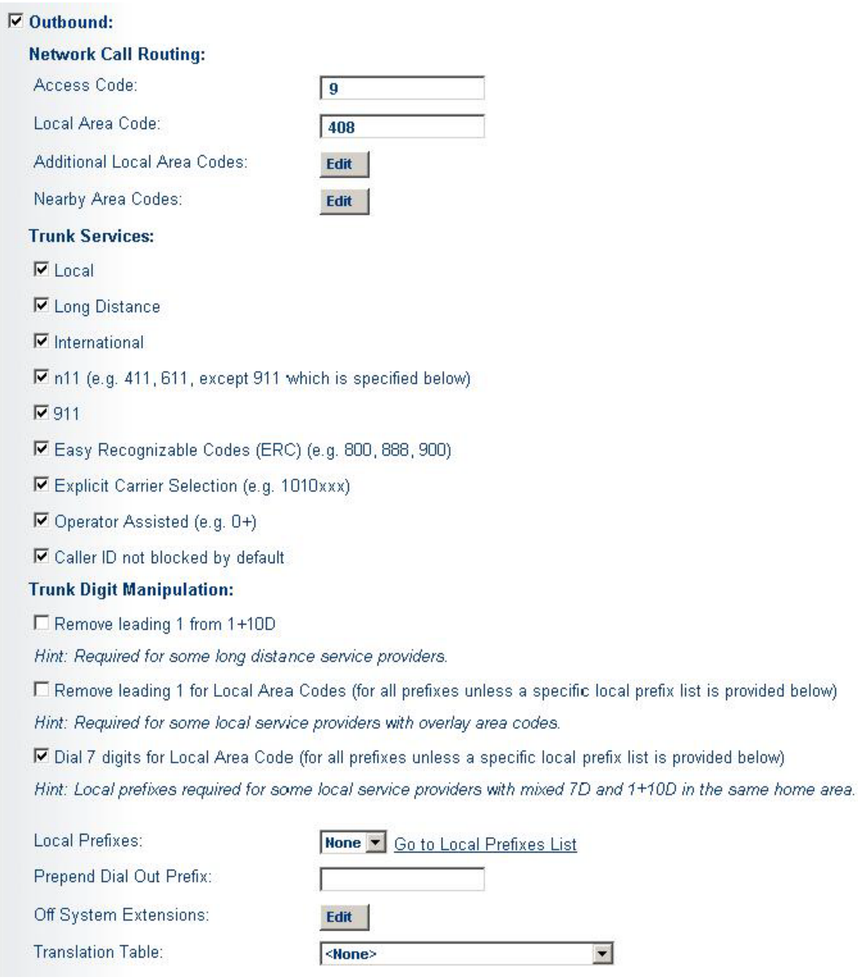


Figure F-3 Creating a SIP trunk group (outbound configurations)

Step 8 Select the **911** check box to enable emergency 911 calls.

You must have at least one trunk group per site that allows 911 calls.

Step 9 Select the **Easy Recognizable Codes (ERC)** check box to enable services such as toll-free dialing calls (e.g., 800, 888, 900).

Step 10 Select the **Explicit Carrier Selection** check box to enable dialing special numbers that let the caller select a long-distance carrier (e.g., 1010xxx).

Step 11 Select the **Operator Assisted** check box to enable the trunk group to dial the operator (e.g., 0+).

Step 12 Select the **Called ID not blocked by default** check box to pass Caller ID information by default on outbound calls. Note that in the United States, the user can override this option with Vertical Service Codes.

Step 13 Click the **Remove leading 1 from 1+10D** check box to drop the leading “1” if your long-distance service provider requires dialing only ten digits.

Step 14 Click the **Remove leading 1 for Local Area Codes** check box to drop the leading “1” for the local area codes (Local and Additional Local) if your local service provider requires dialing only ten digits for local area codes (particularly with overlay area codes).

Step 15 (For all prefixes unless a specific local prefix list is provided below)-Click 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.

Step 16 Click on the **Local Prefixes** drop-down menu and select the local prefix for your site, or click the **Go to Local Prefixes List** link to view, add, and edit the local prefixes for your sites. When you are using a local prefix list, all prefixes not listed are considered "long distance" and calls to these numbers require a long distance trunk service.

Step 17 Enter a prefix in the **Prepend Dial Out Prefix** field to have this prefix prepended to the dial-out string resulting from the other rules. A dial-out prefix is typically required when connecting to, and leveraging the trunks on, a legacy PBX. Note that the Dial Out Prefix is not applied to Off-System Extension calls.

Step 18 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.

F.2.1.3 Create a SIP Trunk

To create a new SIP trunk, follow the procedure below:

Step 1 Launch **ShoreWare 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 **Trunks** link to expand the list (if it has not already been expanded).

Step 4 Click on the **Individual Trunks** link.

Step 5 In the **Add new trunk at** site drop-down menu, select the location where the new SIP trunk will be added. In the **in trunk group** drop-down menu, select the name of the trunk group that you created in the previous task above.

Step 6 Click **Go** to display a window similar to the one shown below.

Step 7 Enter a name for the trunk in the **Name** field.

Step 8 Click on the **Switch** drop-down menu and select the switch that the new trunk will be associated with.

Step 9 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.

Figure F-4 Creating a SIP trunk

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 10 Enter the desired number of SIP trunks in the **Number of Trunks** field.

Step 11 Click the **Save** button to store your changes.

F.2.2 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)

APPENDIX G

Installing and Configuring Reverse Proxy Servers for ShoreTel Communicator for iPhone

ShoreTel Communicator for iPhone can communicate with the ShoreTel UC system via the cellular data network or WIFI. A VPN connection must be used, unless the system uses a configured reverse proxy server.

The data transmitted through the server is not encrypted by default. An option to configure secure communication using SSL is available, and requires an additional reverse proxy server. A VPN connection is not required in this configuration.

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

G.1 Reverse Proxy Settings

Each ShoreTel user is associated with a ShoreTel HQ or DVS Server defined by the user's association to a Site in Director. The Server handles ShoreTel Communicator requests to perform telephony, voicemail, and other actions. In a reverse proxy configuration, a user must use the proxy configuration which will connect the user directly to his Server.

In the case where the user re-assigns his extension to a phone associated to a server different from the server associated with the proxy definition, Communicator for iPhone will not display call history.

The reverse proxy maps public IPs/ports on specific paths with internal IPs/ports and paths. The data received by the Reverse Proxy is routed to the internal ShoreTel Services.

Each user must provision ShoreTel Communicator for iPhone with the appropriate reverse proxy IP address/port.

For example:

User 1 is on HQ at 10.0.0.1

User 2 is on DVS at 10.0.0.2

Reverse proxy is 10.0.0.64 (64.0.0.1 internally) using ports 5500 and 5501

Reverse proxy for User 1 is on HQ could be configured by the administrator:

10.0.0.64:5500/authenticate mapped to 10.0.0.1:80

10.0.0.64:5500/cas mapped to 10.0.0.1:5447

10.0.0.64:5500/director2 mapped to 10.0.0.1:5449

Reverse proxy for User 2 is on DVS could be configured by the administrator:

10.0.0.64:5501/authenticate mapped to 10.0.0.2:80

10.0.0.64:5501/cas mapped to 10.0.0.2:5447

10.0.0.64:5501/director2 mapped to 10.0.0.2:5449

When setting up an account in ShoreTel Communicator, User 1 must use the reverse proxy connection: 10.0.0.64 port: 5500 and User 2 must use the reverse proxy connection: 10.0.0.64 port: 5501.

A single reverse proxy server can be configured to provide services for multiple ShoreTel Services.

In all cases, the IT administrator must make sure that the reverse proxy can be accessed internally and externally.

Note: The reverse proxy configuration uses SSL. A valid SSL certificate signed by a root certificate authority, such as Verisign, must be installed on the reverse proxy server for communication over SSL to be secure.

G.2 Example Reverse Proxy Configuration for Apache 2.2

The following script example illustrates how to configure the reverse proxy on port 5500 for connection to the HQ server at 10.0.0.1.

Step 1 Modify httpd.conf to specify proxy port to be used for HTTP+SSL:

```
#
# Listen: Allows you to bind Apache to specific IP addresses and/or
# ports, instead of the default. See also the <VirtualHost>
# directive.
#
# Change this to Listen on specific IP addresses as shown below to
# prevent Apache from glomming onto all bound IP addresses.
#
Listen 5500
```

Step 2 Verify in httpd.conf that the following modules are enabled (uncommented):

```
LoadModule proxy_module modules/mod_proxy.so
LoadModule proxy_http_module modules/mod_proxy_http.so
LoadModule rewrite_module modules/mod_rewrite.so
LoadModule ssl_module modules/mod_ssl.so
```

Step 3 Edit conf\extra\ httpd-vhosts.conf:

```
#
# Use name-based virtual hosting.
#

NameVirtualHost *:5500

<VirtualHost *:5500>

    # necessary for rewriting
    RewriteEngine on

    # uncomment the logging for problem trace only
    # RewriteLog "logs/ciproxy.localhost-rewrite.log"
    # RewriteLogLevel 3

    # NOTE the rewrite rules have a proxy redirect
    RewriteRule ^/theme/(.+)$ /director2/theme/$1 [P]
    RewriteRule ^/yui_2.7.0/(.+)$ /director2/yui_2.7.0/$1 [P]
    RewriteRule ^/js/(.+)$ /director2/js/$1 [P]

    ProxyPass /authenticate/ http://10.0.0.1/
    ProxyPassReverse /authenticate/ http://10.0.0.1/

    ProxyPass /cas/ http://10.0.0.1:5447/
    ProxyPassReverse /cas/ http://10.0.0.1:5447/

    ProxyPass /director2/ http://10.0.0.1:5449/
    ProxyPassReverse /director2/ http://10.0.0.1:5449/

    # These are Optional
    ErrorLog "logs/ciproxy.localhost-error.log"
    CustomLog "logs/ciproxy.localhost-access.log" combined

    #   SSL Engine Switch:
```

```
#   Enable/Disable SSL AND PROXYING OF SSL for this virtual host.
SSLEngine on
SSLProxyEngine on

#   SSL Cipher Suite:
#   List the ciphers that the client is permitted to negotiate.
#   See the mod_ssl documentation for a complete list.
SSLCipherSuite ALL:!ADH:!EXPORT56:RC4+RSA:+HIGH:+MEDIUM:+LOW:+SSLv2:+EXP:+eNULL

#   Server Certificate:
#   Point SSLCertificateFile at a PEM encoded certificate.  If
#   the certificate is encrypted, then you will be prompted for a
#   pass phrase.  Note that a kill -HUP will prompt again.  Keep
#   in mind that if you have both an RSA and a DSA certificate you
#   can configure both in parallel (to also allow the use of DSA
#   ciphers, etc.)
#SSLCertificateFile "conf/ssl.crt/server-dsa.crt"
SSLCertificateFile "conf/ssl.crt/server.crt"

#   Server Private Key:
#   If the key is not combined with the certificate, use this
#   directive to point at the key file.  Keep in mind that if
#   you've both a RSA and a DSA private key you can configure
#   both in parallel (to also allow the use of DSA ciphers, etc.)
#SSLCertificateKeyFile "conf/ssl.key/server-dsa.key"
SSLCertificateKeyFile "conf/ssl.key/server.key"

#   Server Certificate Chain:
#   Point SSLCertificateChainFile at a file containing the
#   concatenation of PEM encoded CA certificates which form the
#   certificate chain for the server certificate. Alternatively
#   the referenced file can be the same as SSLCertificateFile
#   when the CA certificates are directly appended to the server
#   certificate for convenience.
#SSLCertificateChainFile "conf/ssl.crt/server-ca.crt"

#   Certificate Authority (CA):
#   Set the CA certificate verification path where to find CA
#   certificates for client authentication or alternatively one
#   huge file containing all of them (file must be PEM encoded)
#   Note: Inside SSLCACertificatePath you need hash symlinks
#         to point to the certificate files. Use the provided
#         Makefile to update the hash symlinks after changes.
#SSLCACertificatePath "conf/ssl.crt"
#SSLCACertificateFile "conf/ssl.crt/ca-bundle.crt"

#   Certificate Revocation Lists (CRL):
#   Set the CA revocation path where to find CA CRLs for client
#   authentication or alternatively one huge file containing all
#   of them (file must be PEM encoded)
#   Note: Inside SSLCARevocationPath you need hash symlinks
#         to point to the certificate files. Use the provided
#         Makefile to update the hash symlinks after changes.
#SSLCARevocationPath "conf/ssl.crl"
#SSLCARevocationFile "conf/ssl.crl/ca-bundle.crl"

#   Client Authentication (Type):
#   Client certificate verification type and depth.  Types are
#   none, optional, require and optional_no_ca.  Depth is a
#   number which specifies how deeply to verify the certificate
```

```
#   issuer chain before deciding the certificate is not valid.
#SSLVerifyClient require
#SSLVerifyDepth  10

#   Access Control:
#   With SSLRequire you can do per-directory access control based
#   on arbitrary complex boolean expressions containing server
#   variable checks and other lookup directives.  The syntax is a
#   mixture between C and Perl.  See the mod_ssl documentation
#   for more details.
#<Location />
#SSLRequire (    %{SSL_CIPHER} !~ m/^(EXP|NULL)/ \
#            and %{SSL_CLIENT_S_DN_O} eq "Snake Oil, Ltd." \
#            and %{SSL_CLIENT_S_DN_OU} in {"Staff", "CA", "Dev"} \
#            and %{TIME_WDAY} >= 1 and %{TIME_WDAY} <= 5 \
#            and %{TIME_HOUR} >= 8 and %{TIME_HOUR} <= 20       ) \
#           or %{REMOTE_ADDR} =~ m/^192\.76\.162\.[0-9]+$/
#</Location>

#   SSL Engine Options:
#   Set various options for the SSL engine.
#   o FakeBasicAuth:
#     Translate the client X.509 into a Basic Authorisation.  This means that
#     the standard Auth/DBMAuth methods can be used for access control.  The
#     user name is the `one line' version of the client's X.509 certificate.
#     Note that no password is obtained from the user. Every entry in the user
#     file needs this password: `xxj31ZMTZzkVA'.
#   o ExportCertData:
#     This exports two additional environment variables: SSL_CLIENT_CERT and
#     SSL_SERVER_CERT. These contain the PEM-encoded certificates of the
#     server (always existing) and the client (only existing when client
#     authentication is used). This can be used to import the certificates
#     into CGI scripts.
#   o StdEnvVars:
#     This exports the standard SSL/TLS related `SSL_*' environment variables.
#     Per default this exportation is switched off for performance reasons,
#     because the extraction step is an expensive operation and is usually
#     useless for serving static content. So one usually enables the
#     exportation for CGI and SSI requests only.
#   o StrictRequire:
#     This denies access when "SSLRequireSSL" or "SSLRequire" applied even
#     under a "Satisfy any" situation, i.e. when it applies access is denied
#     and no other module can change it.
#   o OptRenegotiate:
#     This enables optimized SSL connection renegotiation handling when SSL
#     directives are used in per-directory context.
#SSLOptions +FakeBasicAuth +ExportCertData +StrictRequire
<FilesMatch "\.(cgi|shtml|pl|asp|php)$">
    SSLOptions +StdEnvVars
</FilesMatch>
<Directory "C:/xampp/cgi-bin">
    SSLOptions +StdEnvVars
</Directory>

#   SSL Protocol Adjustments:
#   The safe and default but still SSL/TLS standard compliant shutdown
#   approach is that mod_ssl sends the close notify alert but doesn't wait for
#   the close notify alert from client. When you need a different shutdown
#   approach you can use one of the following variables:
#   o ssl-unclean-shutdown:
```

```
    #     This forces an unclean shutdown when the connection is closed, i.e. no
    #     SSL close notify alert is send or allowed to received.  This violates
    #     the SSL/TLS standard but is needed for some brain-dead browsers. Use
    #     this when you receive I/O errors because of the standard approach where
    #     mod_ssl sends the close notify alert.
    #   o ssl-accurate-shutdown:
    #     This forces an accurate shutdown when the connection is closed, i.e. a
    #     SSL close notify alert is send and mod_ssl waits for the close notify
    #     alert of the client. This is 100% SSL/TLS standard compliant, but in
    #     practice often causes hanging connections with brain-dead browsers. Use
    #     this only for browsers where you know that their SSL implementation
    #     works correctly.
    #   Notice: Most problems of broken clients are also related to the HTTP
    #   keep-alive facility, so you usually additionally want to disable
    #   keep-alive for those clients, too. Use variable "nokeepalive" for this.
    #   Similarly, one has to force some clients to use HTTP/1.0 to workaround
    #   their broken HTTP/1.1 implementation. Use variables "downgrade-1.0" and
    #   "force-response-1.0" for this.
    BrowserMatch ".*MSIE.*" nokeepalive ssl-unclean-shutdown downgrade-1.0 force-response-1.0

    #   Per-Server Logging:
    #   The home of a custom SSL log file. Use this when you want a
    #   compact non-error SSL logfile on a virtual host basis.
    #    CustomLog "logs/ssl_request.log" "%t %h %{SSL_PROTOCOL}x %{SSL_CIPHER}x \"%r\" %b"

</VirtualHost>
```

Glossary

Administrator The office manager or IS professional responsible for installing and configuring the system.

All Trunks Busy The situation in which a user tries to make an outside call through a telephone system and receives a “fast” busy signal (twice as many as normal in the same amount of time), indicating that no trunks are available to handle the call.

API Application programming interface; software that an application program uses to request and carry out lower-level services performed by the computer’s or telephone system’s operating system. For Windows, the API also helps applications manage windows, menus, icons, and other graphical user interface elements.

Automated Attendant A device that answers callers with a recording and allows callers to route themselves to an extension; also called an auto-attendant.

BOOTP Boot Protocol, a standard protocol for assigning networking information to client workstations over the network; similar to but less sophisticated than DHCP.

Call Control The dynamic, transactional servicing of calls, usually via a graphical user interface with call information. For example, an attendant can use a GUI application to transfer calls based on CallerID information.

Call Handling The predetermined, preconfigured features for servicing incoming calls in order to obtain certain expected results. Examples of call handling features include call forwarding on busy, call forwarding on no answer, and do not disturb.

Call Handling Mode A set of telephony and call handling features that are enabled depending on the business conditions of the user (for example, in the office or out of the office). Call handling modes, which are enabled manually by the user, include features such as call forwarding on busy, call forwarding on no answer, and the selection of the voice mail greeting to use for a particular mode.

Call History The visual records in ShoreWare Desktop, documenting all incoming and outgoing calls to the user’s extension.

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 methodology 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 Stack The list of calls in ShoreWare Desktop associated with an extension, including active calls and calls that have been put on hold or are being managed in some other way by the user.

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.

Caller For documentation purposes, an outside caller—a person calling the telephone system from outside. See also **End User**.

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.

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.

CLASS Custom Local Area Signalling Services, a family of telephone services offered from local telephone companies, usually for a monthly fee; includes features such as CallerID, Call Waiting, call return, repeat dialing, call rejection, call trace, priority ringing, and selective call forwarding.

Class of Service Abbreviated as CoS or COS; a set of features and privileges associated with a particular user or extension, used for grouping similar users together.

CO Central Office; the building where the telephone company's telephone switching equipment that services the local area is located.

CO Line See **Trunk**.

Conference Three or more parties joined together in a single call, such that each party can hear and be heard by the others.

DHCP Dynamic Host Configuration Protocol, a protocol for downloading network information (such as IP addresses) to client workstations.

DID Direct Inward Dial, a signaling mechanism used by telephone companies to indicate to a customer's PBX what telephone number was dialed by the calling party. It can be used with analog lines but is used mostly with digital (that is, T-1) connections.

DTMF Dual-Tone Multi-Frequency, a technique of providing two tones for each button on a telephone to signal dialing digits; also known as Touch Tone.

End User For documentation purposes, a person using the telephone system from the inside, such as from an extension or a call control application, as opposed to a caller who dials in from outside the system; often shortened to "user." See also **Caller**.

Erlang Formula A mathematical way of predicting a randomly arriving workload (such as telephone calls) based on known information (such as average call duration). Although traditionally used in telephone traffic engineering to determine the required number of trunks, Erlang formulas have applications in call center staffing as well.

External Call A telephone call directed to or from outside the telephone system, and over the Public Switched Telephone Network (PSTN).

FSK Frequency Shift Key, a modulation technique used with low-speed modems; also used with CallerID and message-waiting lamp indicators.

FXO Foreign Exchange Office. 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).

FXS Foreign Exchange Station. 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.

Greeting The voice recording sent to the caller when a call is answered by voice mail or by the auto-attendant; usually a single file, and not the concatenation of smaller phrases.

GUI In ShoreTel documentation, the graphical user interface presented to the user as part of the software application that runs on the user's workstation.

Handled Call A call answered by an employee or a device, such as an auto-attendant or voice mail, as opposed to being blocked or abandoned.

Hang Up The act of putting the telephone receiver back on the hook to indicate to the telephone system that the user is done with the call.

Hold As in "on hold"; the situation in which a caller is placed in the user's call management stack for later handling.

Internal Call A telephone call dialed between internal extensions.

Java The platform-independent programming language developed by Sun Microsystems for providing complete programs, including animated graphics.

Line See **Trunk**.

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.

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.

On Hook/Off Hook The state of the telephone as being either on the hook (hung up) or off the hook and seizing the line.

Operator The person who monitors the telephone system and transfers calls to the appropriate extensions.

Outside Caller See *Caller*.

PBX Private Branch Exchange; a term used by telephone companies to indicate equipment that is located on the customer's premises and that can route telephone calls.

Permissions Privileges granted to each user with respect to what data, features, menus, or calling options may be used. Permissions are under the control of the system administrator.

Physical Extension A common internal extension with an assigned physical port and telephone.

Prompt For an auto-attendant menu, the result of playing (concatenating) a series of phrases together.

PSTN Public Switched Telephone Network; another name for the public telephone network.

Remote Caller See *Caller*.

Ringback Tone The audible signal given to the caller by the telephone company (or telephone system) to indicate that the remote telephone is ringing.

RJ-11 Registered Jack number 11; one of the series of registered jacks and cabling developed originally by AT&T to standardize the cabling between the telephone and the telephone company lines.

Service Provider Interface (SPI) An interface between the operating system and the telephone hardware.

Status Bar A text and mini-graphics area, usually at the bottom of a software application window, that is normally used for showing the status of the application or other pertinent information.

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.

T-1 A digital transmission link with a capacity of 1.554 Mbps (1,544,000 bits per second). 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.

Telco An abbreviation for telephone company.

Telephony Application Programming Interface (TAPI) A telephony software interface included in Microsoft Windows 95, 98, and NT; the operating system that lets applications incorporate telephony control.

Tip and Ring Telephony jargon for the two wires from the telephone system to the telephone set; also indicates polarity

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.

TUI Telephone User Interface; a set of defined keystrokes on the telephone keypad that are used to execute commands to either the telephony switch, voice mail, or the automated attendant.

Workstation A personal computer (PC) or similar computer.

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