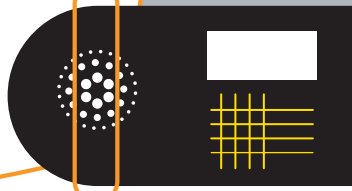




PRODUCT DOCUMENTATION

ShoreTel
Administration
Guide

RELEASE 8.1



Document and Software Copyrights

Copyright © 1998–2008 by ShoreTel, Inc., Sunnyvale, California, U.S.A. All rights reserved. Printed in the United States of America. Contents of this publication may not be reproduced or transmitted in any form or by any means, electronic or mechanical, for any purpose, without prior written authorization of ShoreTel, Inc.

ShoreTel, Inc. reserves the right to make changes without notice to the specifications and materials contained herein and shall not be responsible for any damage (including consequential) caused by reliance on the materials presented, including, but not limited to, typographical, arithmetic, or listing errors.

Trademarks

ShoreTel[®], ShoreCare[®], ShoreWare[®], ShoreGear[®] and ControlPoint[®] are registered trademarks of ShoreTel, Inc. in the United States and/or other countries. ShorePhone[™] is a trademarks of ShoreTel, Inc. in the United States and/or other countries. All other copyrights and trademarks herein are the property of their respective owners.

Patents

This product is covered by one or more of the following patents: United States patent 6,996,059, United States patent 7,003,091, United States patent 7,167,486, United States patent 7,379,540, and United States patent 7,386,114. ShoreTel, Inc. All rights reserved.

Version Information

ShoreTel 8.1 Administration Guide
Revision 3
Part Number: 800-1227-01
Date: November 5, 2008

Company Information

ShoreTel, Inc.
960 Stewart Drive
Sunnyvale, California 94085
(408) 331-3300
(408) 331-3333 fax
www.shoretel.com

Table of Contents

TABLE OF CONTENTS	III
PREFACE	I
CHAPTER 1: USING SHOREWARE DIRECTOR	1
1.1 Introduction	1
1.2 Overview of ShoreWare Director	1
1.2.1 Architectural Overview	1
1.2.2 Multi-level Management	2
1.2.3 Multi-user Management	2
1.2.4 External Telephone Number Formatting	2
1.2.5 Preferences	3
1.2.6 Prompts	5
1.3 Login	5
1.4 Product Registration	6
1.5 Navigation Frame	7
1.5.1 Getting Help	7
1.5.2 Logging Off	8
1.6 Data-Entry Frame	8
1.6.1 List Panels	8
1.6.2 Edit Panels	10
1.7 Getting Started	11
CHAPTER 2: SETTING UP SYSTEM PARAMETERS	15
2.1 Overview	15
2.2 Dialing Plan	16
2.2.1 Parameters	16
2.2.2 Increasing Extension Length	18
2.3 Digit Translation Tables	18
2.4 System Extensions	20
2.4.1 Parameters	20
2.5 SNMP	21
2.5.1 Parameters	22
2.6 BOOTP Server	22
2.7 Other Parameters	23
2.7.1 Parameters	23
2.8 Languages	24
2.9 Licenses	25
2.9.1 Compliance	25
2.9.2 Node Locking – System key	26
2.9.3 Keyed License Types	26
2.9.4 Extension and Mailbox Licenses	28
2.9.5 Keyed License Details	29
2.9.6 Additional Language Licenses	29

2.9.7	Self-Audited Licenses	29
2.10	Contact Information	30
2.10.1	Parameters	31
2.10.2	Sending Contact Information	32
2.11	Administrative Permissions	32
2.11.1	Parameters	33
2.11.2	Assigning an Administrative Role to a User	36
CHAPTER 3: SHORETEL SITES		39
3.1	Overview	39
3.2	Configuring Sites	39
3.3	Parameters	40
CHAPTER 4: CONFIGURING APPLICATION SERVERS		45
4.1	Overview	45
4.2	Distributed Voice Mail	46
4.2.1	IP Phone Limitations/Requirements	46
4.2.2	Call Manager Limitations/Requirements	47
4.3	Legacy Voice Mail Integration	47
4.4	Adding Application Servers	48
4.4.1	Parameters	48
4.4.2	Extension List Mapping	52
4.5	Fax Servers	53
CHAPTER 5: CONFIGURING SWITCHES		55
5.1	Switch Models	55
5.1.1	ShoreGear 1-U Half Width Voice Switches	55
5.1.2	ShoreGear 1-U Full Width Voice Switches	56
5.1.3	ShoreGear IPBX Voice Switches	56
5.2	Switch Resources	56
5.2.1	Analog Circuits	56
5.2.2	Digital Circuits	57
5.2.3	IP Phone Ports	57
5.2.4	SIP Trunks	57
5.2.5	SIP Proxies	57
5.3	Configuration Parameters	58
5.3.1	IP Phone, SIP, and Make Me Conference Support	58
5.3.2	Backup Operator	58
5.4	Director Panels	58
5.4.1	Switches List Panel	58
5.4.2	Switch Edit panel	60
5.5	Switch Parameters	62
5.5.1	System Parameters	62
5.5.2	T1 Signalling Parameters	64
5.5.3	E1 Signalling Parameters	65
5.5.4	BRI Signalling Parameters	66
5.5.5	Analog Port Tables	67
5.5.6	Digital Channel Tables	68
5.6	Adding a ShoreGear Switch	70
5.7	SoftSwitch	71

5.8	Replacing a ShoreGear Voice Switch	71
CHAPTER 6: VOICEMAIL MODEL SWITCHES		73
6.1	Introduction	73
6.1.1	Definition	73
6.1.2	Background	73
6.2	Functional Description	74
6.2.1	Physical Description	74
6.2.2	Voice Switch Functions	75
6.2.3	Server Functions	75
6.2.4	Connectivity Requirements	76
6.3	Utilities	76
6.3.1	Accessing Voicemail Model Switch Utilities	76
6.3.2	Switch Utilities	78
6.4	V Model Switch Implementation	79
6.4.1	Installing a Switch	79
6.4.2	Replacing a Switch	80
6.4.3	Upgrading a Switch	81
6.4.4	Configuring Voice Mail	82
6.4.5	Specifying Root and Administrator Passwords	82
6.4.6	Voicemail Backup	84
6.4.7	Configuring a Target Server for Backup	85
6.5	Booting and Restarting	87
6.5.1	Description	87
6.5.2	Specifying a Static IP Address	89
6.5.3	Specifying a Time Source	89
6.5.4	Reboot Methods	90
6.6	Monitoring a Voicemail Model Switch	90
6.6.1	Monitoring Switch Functions	90
6.6.2	Log File Size and Age	91
6.6.3	Monitoring CF Memory Usage	91
CHAPTER 7: CONFIGURING TRUNKS		95
7.1	Overview	95
7.2	Setting Up Trunks	95
7.2.1	Add or Edit a Trunk Group	96
7.2.2	Importing Local Prefixes	101
7.2.3	Edit DID Range	103
7.2.4	Edit DNIS Digit Map	103
7.2.5	Additional Local Area Codes	104
7.2.6	Nearby Area Codes	105
7.2.7	Prefix Exceptions	105
7.2.8	Off-System Extensions	105
7.2.9	Tandem Trunking	107
7.2.10	Centrex Flash	107
7.2.11	Pause in Dialing – Trunk Access	107
7.3	Individual Trunks	109
7.3.1	Add or Edit a Trunk	109
CHAPTER 8: CONFIGURING IP PHONES		111
8.1	Overview	111

8.2	Setting IP Address Ranges	111
8.3	IP Phone Address Map	112
8.3.1	Adding Teleworking IP Phones	112
8.4	Setting IP Phone Options	113
8.5	Adding Individual IP Phones Users	115
8.5.1	Adding Users with Any IP Phone	115
8.5.2	Assigning IP Phones by User	116
8.6	Viewing and Editing IP Phones on the System	117
8.6.1	Viewing IP Phones	117
8.6.2	Renaming IP Phones	118
8.6.3	Deleting and Moving IP Phones	118
8.6.4	IP Phone State Display	119
8.6.5	Displaying ShorePhone Settings	119
8.6.6	Resetting the ShorePhone	119
8.7	Loading Custom Ring Tones	120
8.7.1	Creating the Phone-Specific Configuration File	121
8.8	Customizing Color LCD Displays with Wallpaper	122
8.8.1	Wallpaper File Specifications	122
8.8.2	Downloading the File to Several Color-Screen IP Phones	122
8.8.3	Downloading the File to a Single Color-Screen IP Phone	123
8.9	VPN Phone	124
8.9.1	Introduction	124
8.9.2	Functional Description	125
8.9.3	Implementation	126

CHAPTER 9: SETTING CALL CONTROL OPTIONS 133

9.1	Overview	133
9.2	Account Codes	133
9.2.1	Adding and Editing Account Codes	135
9.3	Bridged Call Appearances	137
9.3.1	Switch Support for Bridged Call Appearances	138
9.3.2	Configuring BCA Parameters	138
9.3.3	Configuring Answer Options	140
9.4	Hunt Groups	141
9.4.1	Setting the Hunt Group to Busy	144
9.5	Paging Groups	145
9.5.1	Adding Overhead Paging to Paging Groups	145
9.5.2	Configuration	145
9.6	Pickup Groups	146
9.6.1	Configuring Pickup Groups	147
9.7	Route Points	149
9.7.1	Parameters	149
9.8	Call Control Options	152
9.8.1	Distributed Routing Service	152
9.8.2	Parameters	154
9.9	Bandwidth Management and Codec Negotiation	156
9.9.1	Codec Lists	156
9.9.2	Intersite Video Sessions	157
9.9.3	Codec Negotiation	157
9.9.4	Implementation	158
9.10	Automatic Ringdown	162

9.10.1	Dedicated Circuit Ringdown	163
9.10.2	Phone Delayed Ringdown	165
9.10.3	Implementation	166
9.11	Media Encryption	171
9.11.1	Description	171
9.11.2	Implementation	172
 CHAPTER 10: CONFIGURING USERS		 173
10.1	Overview	173
10.2	Classes of Service	174
10.2.1	Telephony Features Permissions	175
10.2.2	Whisper Page	179
10.2.3	Call Permissions	180
10.2.4	Voice Mail Permissions	183
10.3	User Groups	185
10.3.1	User Group List Page Column Definitions	185
10.3.2	User Group Parameters	186
10.4	Individual Users	187
10.4.1	Individual Users List Page Column Headings	187
10.4.2	Private Numbers	188
10.4.3	User Import Tool	189
10.4.4	General Information	192
10.4.5	Personal Options	199
10.4.6	Voice Mail Delivery and Notifications	203
10.4.7	Find Me and External Assignment	205
10.4.8	Programmable IP Phone Buttons	207
10.4.9	Programming Personal Call Manager Toolbar	217
10.4.10	Wireless Headset Hook Switch	224
10.4.11	System Directory Record	226
10.4.12	Distribution Lists	227
10.4.13	Workgroups	227
10.5	Notify Users	228
10.6	Anonymous Telephones	229
10.7	Extension Lists	230
10.7.1	Paging Through a List	231
10.8	Batch Update Utility	231
10.8.1	Batch Update	232
10.9	Call Handling Mode Defaults	234
10.9.1	Parameters	234
10.10	Escalation Profiles and Other Mailbox Options	236
10.10.1	Parameters	236
10.10.2	Configuring Voice Mailbox Full Notifications	238
10.10.3	Configuring Escalation Notification	239
 CHAPTER 11: CONFIGURING CLIENT APPLICATIONS		 245
11.1	Call Manager	245
11.1.1	Introduction	245
11.1.2	Installation	245
11.1.3	Configuring Instant Messaging	246
11.1.4	Presence	249
11.1.5	Setting the Maximum Contact List Size	250
11.1.6	Enable SoftPhone for a User	251

11.1.7	Video Calls	252
11.2	Contact Center Integration with Call Manager	253
11.2.1	Description	253
11.2.2	Accessing Contact Center Agent Toolbar Functions	254
11.2.3	Configuration	255
11.2.4	Configuring Call Center Options through Call Manager	257
11.3	Mobile Call Manager	257
11.3.1	Supported Configurations	258
11.3.2	ShoreWare Mobile Call Manager Support – Devices	259
11.3.3	Requirements	260
11.3.4	Administrator Configuration	261
11.3.5	ShoreWare Director Configuration	264
11.3.6	Client Configuration Information Delivery	265
11.3.7	Configuring Information upon First-Time Use	265
11.3.8	Upgrading Mobile Call Manager	268
11.4	Configuring Conference Bridge	270
11.4.1	Adding a Conference Bridge Server	271
 CHAPTER 12: CONFIGURING VOICE MAIL		 273
12.1	System Distribution Lists	273
12.1.1	Parameters	274
12.2	AMIS Voice Mail	275
12.2.1	AMIS Restrictions	276
12.2.2	Enabling AMIS	276
12.3	Voice Mail Options	277
12.3.1	Creating AMIS Systems	277
12.3.2	Disabling AMIS Systems	279
12.3.3	Setting Voice Mail User Group Permissions	279
12.3.4	AMIS Test Mailbox	279
12.4	Voice Mail Reports	280
12.4.1	Summary Report	280
12.4.2	Server Report	281
 CHAPTER 13: CONFIGURING THE AUTO-ATTENDANT		 285
13.1	Multiple Auto-Attendants	285
13.2	Menus	285
13.2.1	Parameters	285
13.2.2	Adding and Editing an Auto-Attendant Menu	286
13.2.3	Configuring an Auto-Attendant Menu	290
 CHAPTER 14: CONFIGURING SCHEDULES		 293
14.1	Overview	293
14.2	On-Hours	294
14.2.1	Configuring the On-Hours Schedule	294
14.3	Holiday Schedule	296
14.3.1	Parameters	296
14.3.2	Configuring the Holiday Schedule	297
14.4	Custom Schedule	297
14.4.1	Parameters	297
14.4.2	Configuring a Custom Schedule	298

CHAPTER 15: CONFIGURING WORKGROUPS	299
15.1 Overview	299
15.1.1 Call Routing	299
15.1.2 Call Distribution	299
15.1.3 Call Queuing	300
15.1.4 Workgroup Call Managers	300
15.1.5 Intercom/Paging, Barge In, Record, and Monitor	301
15.1.6 Reporting	304
15.2 Workgroup Navigation	304
15.3 Adding and Editing Workgroups	305
15.3.1 Parameters	306
15.4 Selecting a DNIS Trunk Group for DNIS Routing	309
15.5 Editing Workgroup Membership	310
15.6 Editing Workgroup Queue Handling	310
15.6.1 Configuring Workgroup Overflow / Interflow	313
15.7 Workgroup Thresholds	314
 CHAPTER 16: MANAGING THE SYSTEM DIRECTORY	 315
16.1 System Directory	315
16.1.1 System Directory Entry Edit Page	316
16.1.2 Data-Entry Fields	316
 CHAPTER 17: SESSION INITIATION PROTOCOL	 319
17.1 SIP Trunks	319
17.1.1 Supported RFCs	320
17.1.2 General SIP Comments	320
17.1.3 Configuration SIP Trunks	323
17.2 SIP Extensions	330
17.2.1 Network Elements	330
17.2.2 Supporting SIP Devices	331
17.2.3 User Features	332
17.2.4 System Features	335
17.2.5 User Assignment	336
17.2.6 Configuration	336
 CHAPTER 18: MAINTENANCE	 345
18.1 Overview	345
18.2 Quick Look	346
18.2.1 Maintenance Switches Summary	347
18.3 Switch Maintenance	350
18.3.1 Switch Section	350
18.3.2 Link Section	351
18.3.3 Ports Section	353
18.3.4 Hunt Groups Section	354
18.3.5 IP Phone Maintenance	355
18.3.6 SIP Trunks Section	355
18.3.7 Details Section	355
18.3.8 ShoreGear-T1 PRI and ShoreGear-E1 PRI Maintenance	357
18.3.9 Quick Look Server Maintenance—Main Server	357
18.3.10 Quick Look Server Maintenance—Distributed Servers	360

18.4	Switch Connectivity	360
18.5	Conference Ports	362
18.6	Event Log	362
	18.6.1 Viewing Event Reports	363
18.7	Services	365
	18.7.1 Service Maintenance Page	365
18.8	Event Filters	365
	18.8.1 Editing and Creating Event Filters	367
18.9	Database Maintenance	368
	18.9.1 MySQL Services	368
	18.9.2 MySQL Backup and Restore Utilities	369
CHAPTER 19: SYSTEM RECOVERY		373
19.1	Introduction	373
19.2	Overview	373
	19.2.1 Back Up	373
	19.2.2 Restore	374
19.3	Process Requirements	374
	19.3.1 Protected Server Files	374
	19.3.2 Location of Script Files	375
	19.3.3 Backup File Location	375
	19.3.4 Main Scripts	375
	19.3.5 Convenience Scripts	376
	19.3.6 Support Files	376
	19.3.7 Estimated Completion Time	376
	19.3.8 Modifications for Non-Standard Installations	377
19.4	Procedure	377
	19.4.1 Setting Up an Alternate Storage Folder	377
	19.4.2 Establishing a Connection to a Voicemail Model Server	378
	19.4.3 Backing up Server Files	379
	19.4.4 Restoring a Server	381
	19.4.5 Log Files	382
CHAPTER 20: REPORTING		383
20.1	Overview	383
20.2	Reports	384
	20.2.1 Reports	384
	20.2.2 Parameters	385
	20.2.3 Converting Existing CDR Data to MySQL Format	386
20.3	Options	387
	20.3.1 Configuring Send CDR Out SMDR Interface	388
	20.3.2 Creating the Archive Database	389
APPENDIX A: USING OFFICE ANYWHERE		391
A.1	Special Considerations	392
A.2	Terminology	393
A.3	Configuring Permission Settings	393
A.4	Configuring Off-Net External Assignment	395
A.5	Configuring On-Net Extension Assignment	397
	A.5.1 Using the Telephone User Interface	397

A.5.2	Using the ShoreTel Call Manager	398
APPENDIX B: EMERGENCY 911 OPERATIONS		399
B.1	Overview	399
B.1.1	How 911 Calls Work	399
B.1.2	Roles and Responsibilities	400
B.1.3	Selecting a PS/ALI Service Provider	401
B.2	Feature Operation	401
B.2.1	Digit Collection for 911 Calls	401
B.2.2	Ensuring Proper Routing of 911 Calls	402
B.2.3	Trunk Signaling for 911 Calls	402
B.3	Selecting Caller ID Type for 911 Calls	403
B.3.1	Available Caller ID Options	403
B.4	Configuring Your System	405
B.4.1	Trunk Groups	407
B.4.2	User Groups	407
B.4.3	Users	408
B.4.4	IP Phone Address Map	409
B.4.5	Switch	409
B.4.6	Sites	409
B.5	Planning Your Emergency Response	410
B.5.1	Call Notification	410
B.6	International Emergency Numbers	411
B.6.1	Special Considerations for Netherlands	412
B.7	Verifying Your 911 Configuration	412
B.8	Additional Recommendations	412
APPENDIX C: CALL DETAIL RECORD REPORTS		415
C.1	Overview	415
C.1.1	Call Accounting Service	415
C.1.2	Active CDR Database	416
C.1.3	Legacy CDR Text Files	416
C.2	MySQL Database	417
C.2.1	Compatibility and Pre-Configuration Requirements	418
C.2.2	Archival and Backup Utilities	418
C.2.3	Installing MySQL on a Secondary Server	421
C.2.4	Importing Access Database Files into MySQL Format	422
C.2.5	Performance Tuning for Report Generation	423
C.2.6	Report generation time – CPU Utilization	424
C.2.7	MySQL CDR Database and Internationalization	424
C.2.8	Monitor MySQL service	424
C.2.9	Tools for browsing MySQL database tables	424
C.2.10	Restrictions in the number of records returned by the MySQL CDR query	424
C.2.11	Report License Error Condition	424
C.3	CDR Reports	425
C.3.1	TMS-CDR Media Stream Statistics	426
C.3.2	Generating CDR Reports	429
C.4	Interpreting CDR Reports	429
C.4.1	User Activity Summary Report	429
C.4.2	User Activity Detail Report	431
C.4.3	Trunk Activity Summary Report	433
C.4.4	Trunk Activity Detail Report	434

C.4.5	Workgroup Agent Summary Report	436
C.4.6	Workgroup Agent Detail Report	437
C.4.7	Workgroup Service Level Summary Report	438
C.4.8	Workgroup Queue Summary Report	442
C.4.9	Media Stream Summary Report	444
C.4.10	Media Stream Detail Report	445
C.4.11	Account Code Summary Report	447
C.4.12	Account Code Detail Report	447
C.5	CDR Database	447
C.5.1	Call Table	450
C.5.2	Enumeration Tables: Use for the Call Table	452
C.6	Connect Table	453
C.6.1	Enumeration Tables Used for Connect Table	453
C.7	MediaStream Table	456
C.8	AgentActivity Table	457
C.8.1	Enumeration Tables Used for AgentActivity	458
C.9	QueueCall Table	459
C.9.1	Enumeration Tables Used for QueueCall	460
C.10	QueueStep Table	461

CHAPTER D: SHOREGEAR SWITCHES 463

D.1	Switch Models	463
D.1.1	ShoreGear 1-U Half Width Voice Switches	463
D.1.2	ShoreGear Voicemail Model Voice Switches	464
D.1.3	ShoreGear 1-U Full Width Voice Switches	465
D.1.4	ShoreGear IPBX Voice Switches	465
D.2	Specifications – SG 1-U Half-Width Switches	466
D.2.1	ShoreGear 90 Voice Switch	466
D.2.2	ShoreGear 90BRI Voice Switch	469
D.2.3	ShoreGear 50 Voice Switch	472
D.2.4	ShoreGear 30 Voice Switch	475
D.2.5	ShoreGear 30BRI Voice Switch	479
D.2.6	ShoreGear 220T1 Voice Switch	482
D.2.7	ShoreGear 220T1A Voice Switch	485
D.2.8	ShoreGear-220E1 Voice Switch	488
D.2.9	ShoreGear T1k Voice Switch	491
D.2.10	ShoreGear E1k Voice Switch	494
D.3	Specifications – SG Voice Model Switches	498
D.3.1	ShoreGear 90V Voice Switch	498
D.3.2	ShoreGear 90BRIV Voice Switch	501
D.3.3	ShoreGear 50V Voice Switch	504
D.4	Specification – SG 1U Full Width Switches	508
D.4.1	ShoreGear 120 Voice Switch	508
D.4.2	ShoreGear 24A Voice Switch	511
D.4.3	ShoreGear 60 Voice Switch	514
D.4.4	ShoreGear-T1 and ShoreGear E1 Voice Switches	521
D.5	Specification – IPBX Switches	524
D.5.1	IPBX-24 Voice Switch	524
D.5.2	IPBX-T1 and IPBX-E1 Voice Switch	527

INDEX

531

Preface

This preface provides information about the objectives, organization, and conventions of the *ShoreTel 8.1 Administration Guide*.

Objectives

This guide explains how to use ShoreWare Director to configure, administer, and maintain the *ShoreTel 8.1* system.

The planning and installation procedures are covered in the *ShoreTel 8.1 Planning and Installation Guide*.

Audience

This guide is written for the person who uses ShoreWare Director to administer and maintain the *ShoreTel 8.1* system.

Organization

This guide is organized to reflect the order in which you initially configure the *ShoreTel 8.1* system.

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

Documentation Overview

The *ShoreTel 8.1* system is documented as described in the following sections.

System Documentation

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

- *ShoreTel 8.1 Planning and Installation Guide* provides information on how to plan the implementation of the *ShoreTel 8.1* system, as well as how to install the necessary hardware, data communications, and telecommunications elements. The *ShoreTel 8.1 Planning and Installation Guide* can be used in conjunction with the ShoreCare ControlPoint project management tool.
- *ShoreTel 8.1 Administration Guide* (this guide) provides detailed information on how to administer and maintain the *ShoreTel 8.1* system using ShoreWare Director. This includes task-based information, as well as screen-by-screen information regarding ShoreWare Director.

Hardware Documentation

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

- *ShoreGear Quick Install Guide*
- *ShoreTel IP Phone Quick Install Guide*

If you have also purchased one or more ShoreTel Converged Conference Bridges, refer to the *ShoreTel Conference Bridge Quick Install Guide* and the *ShoreTel Converged Conference Solution Administration Guide*.

If you have purchased the ShoreTel Contact Center Solution, refer to the *ShoreTel Contact Center Solution Administration Guide* and the *ShoreTel Contact Center Solution Planning and Installation Guide*.

User Documentation

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

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

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 release or to resolve issues with the product, use the ShoreTel Communications Online Knowledge Base, accessible from the ShoreTel web site at www.shoretel.com.

Document Conventions

The following conventions are used in this guide:

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

Using ShoreWare Director

1.1 Introduction

This chapter provides information about using ShoreWare Director. Sections in this chapter include:

- Overview of ShoreWare Director
- Login
- Product Registration
- Navigation Frame
- Data-Entry Frame
- Getting Started

1.2 Overview of ShoreWare Director

ShoreWare Director is a web-based administration and maintenance tool that lets you manage your ShoreTel 6.1 system from anywhere on your IP network. The ShoreTel 6.1 system has a unique distributed call control architecture as well as a suite of voice applications that provides a single image system for all users across all sites. Gone are the days of having PBXs, voice mail systems, automated attendants, or ACD systems, each with their own dedicated management interface. ShoreWare Director lets you manage, from a single web management interface, all your users and trunks, including the call control features, and all the voice applications (voice mail, automated attendant, workgroups, call detail recording, unified messaging, and desktop call control).

1.2.1 Architectural Overview

The main ShoreWare server hosts the web site for ShoreWare Director using Microsoft Internet Information Server. When you launch a web browser and navigate to the ShoreWare Director web site, the server provides HTML web pages from which you can add to, delete from, and edit the configuration of the system (see Figure 1-1). When you click **Save**, your change is sent to the server and saved in the ShoreTel 6.1 database. All other system components are automatically and immediately notified and updated.

In addition to the configuration panels, ShoreWare Director has a maintenance interface for the ShoreTel 6.1 system. When you navigate to a maintenance panel in ShoreWare Director, system status is displayed, and you can issue maintenance commands. The commands are passed to and executed by the server. If you have distributed ShoreWare application servers, you can navigate from the main server to each distributed server through ShoreWare Director to view status and issue commands to the distributed server.

ShoreWare Director is supported only by Microsoft Internet Explorer 5.5 and later versions. If you are running ShoreWare Director from Windows XP (SP2), the Director site (i.e. URL) must be added to Internet Explorer's list of "trusted sites."

First Name	Last Name	Site	User Group	Client Type	Extension	Mailbox	Switch	Port	Status
Adam	Ant	Headquarters	Executives	Operator	4509	4509	Doc-24	ShoreDoc IP Phone	Home
Betty	Boop	Headquarters	Executives	Operator	4510	4510	Doc-24	10	Home
Carl	Castle	Headquarters	Executives	Operator	4511	4511	Doc-24	11	Home
Donald	Duck	Headquarters	Executives	Operator	4512	4512	ShoreDoc-HQ	Any IP Phone	Home
Ernie	Els	Headquarters	Executives	Personal	4513	4513	Doc-24	13	Home
Fred	Flintstone	Headquarters	Executives	Personal	4514	4514	Doc-24	14	Home
Greta	Garbo	Headquarters	Executives	Personal	4515	4515	Doc-24	15	Home
Howard	Hughes	Headquarters	Executives	Personal	4516	4516	Doc-24	16	Home
Johnathan	Seagull	Headquarters	Executives	Personal	4805	4805	Doc-24	1	Home
Tim	Toolman	Headquarters	Executives	Personal	4904	4904	Bedrock	10	Home
Ursula	Urp	Headquarters	Executives	Personal	4907	4907	Bedrock	13	Home
Vince	Vinceble	Headquarters	Executives	Personal	4906	4906	Bedrock	12	Home
Wanda	Wenderly	Headquarters	Executives	Personal	4908	4908	Bedrock	14	Home
Wendy	Williams	Headquarters	Executives	Personal	4905	4905	Bedrock	11	Home
Yourst	Yumbal	Headquarters	Executives	Personal	4903	4903	Bedrock	9	Home

© 1998-2004 ShoreTel, Inc. All rights reserved.

Figure 1-1 ShoreWare Director Individual Users List Panel

1.2.2 Multi-level Management

The ShoreTel 6.1 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 panels, you can define site administrators, directory list managers, read-only users, and more. Each user who needs to access Director can be assigned a level of permission tailored for his needs.

1.2.3 Multi-user Management

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

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

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

1.2.4 External Telephone Number Formatting

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

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

- +C... (A...) S...
 - “+” = International designation
 - “C” = Country code
 - “A” = Area code (also known as a city code)
 - “S” = Subscriber number

DID Numbers – Must be entered in canonical format.

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

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

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

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

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

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

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

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

Table 1-1 gives example phone numbers for all the countries supported by ShoreWare Director.

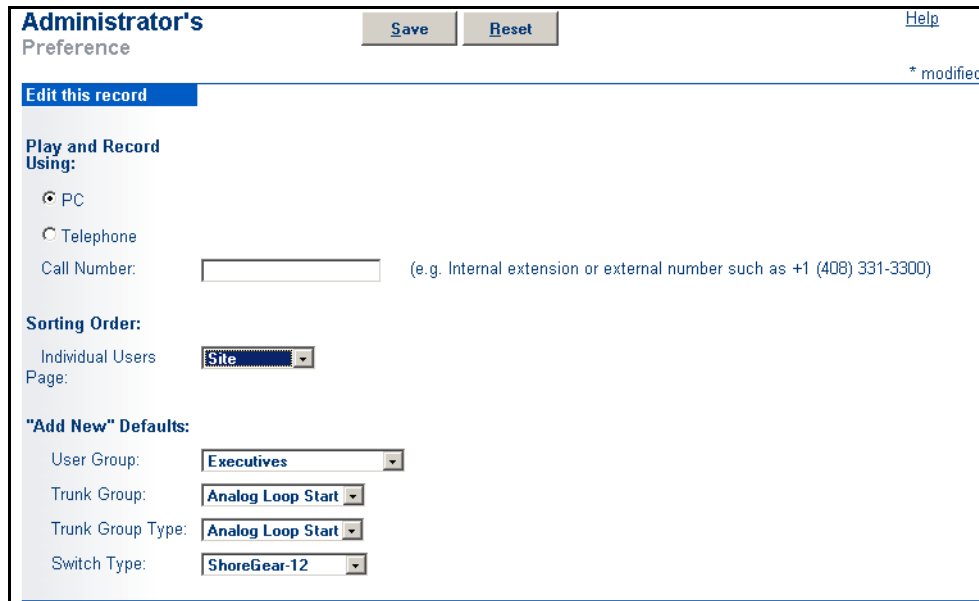
Phone Number	Country
+1(408) 331-3300	U.S., Canada
+31 70 348 6486	Netherlands
+33 8 36 68 31 12	France
+34 91 845 6078	Spain
+44 20 7634 8700	UK
+49 69 571903	Germany
+55 61 429 7777	Brazil
+60 3 2693 5188	Malaysia
+61 2 9360 1111	Australia
+65 736 6622	Singapore
+852 2508 1234	Hong Kong

Table 1-1 International Phone Number Examples

1.2.5 Preferences

ShoreWare Director saves certain preferences in “cookies” on the client PC for ease of use. The panel shown in Figure 1-2 lets you configure some of these preferences, including how you record auto-attendant prompts and exposes some defaults. To view this panel, click *Preferences* under the *Administration* link in the navigation frame (described later in this chapter).

The parameters that you can set on the panel shown in Figure 1-2 are described here:



Administrator's Preference Save Reset [Help](#)

* modified

Edit this record

Play and Record Using:

PC

Telephone

Call Number: (e.g. Internal extension or external number such as +1 (408) 331-3300)

Sorting Order:

Individual Users:

Page:

"Add New" Defaults:

User Group:

Trunk Group:

Trunk Group Type:

Switch Type:

Figure 1-2 Preference Edit Panel

- **Play and Record Using:** This controls how you play and record an auto-attendant prompt.
 - *PC* – Click this button if you will play and record via your PC's speakers and microphone. This requires that you have a sound card installed in your computer.
 - *Telephone* – Click this button if you will play and record via your telephone handset. In the *Call Number* field, you must enter a telephone number or the extension at which the system will call you.
- **Sorting Order:** This controls the sort order of the users on the *Individual User List* panel. The column name you select from the *Individual Users* panel drop-down list defines the column that is used to sort the user list in ascending order.
- **"Add New" Defaults:** This defines the default when adding new records. It includes:
 - User Group
 - Trunk Group
 - Trunk Group Type
 - Switch Type

In addition to the preferences that are exposed explicitly, ShoreWare Director will remember other preferences as cookies and leverage them to predict good defaults when adding new records. The following information is also stored locally in cookies:

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

1.2.6 Prompts

Prompts on the ShoreTel 6.1 system can be imported into the system using μ -law, WAV file format (CCITT μ -law, 8KHz, 8 bit, mono). If you want your prompts to match the voice of the ShoreTel 6.1 system, contact Worldly Voices at the web site www.worldlyvoices.com. Request that “Connie” record your prompts. Worldly Voices provides this service for a nominal fee and provides rapid turnaround.

1.3 Login

To access ShoreWare Director, you must enter its Uniform Resource Locator (URL) in your web browser's address field. Typically, this has the form:

```
http://<shoreware_server_name>/ShoreWareDirector
```

After entering the URL, click Go or press the Enter key, and the login panel appears as shown in Figure 1-3. The build number for the ShoreWare software appears above the copyright number.

You can use the name of your ShoreWare server if you have a name resolution service available; otherwise, use the IP address of the server.



Figure 1-3 Login panel

To log in to ShoreWare Director:

- Step 1** Enter your user ID and password in the User ID and Password fields. The default user ID and password for the system are `admin` and `changeme`.

The admin and changeme defaults must be changed once you assign full System Administrator permission to a user. See the Section 2.11 on page 32 for information.

Step 2 Click *Login* to log in or click *Cancel* to re-enter your user ID and password.

You can find the current build number associated with the installed software in the lower left corner of the Login Panel.

1.4 Product Registration

After you have logged into the system, the product registration panel, shown in Figure 1-4, appears, providing you with the following registration options:

- **Now:** Takes you to the *System Parameters Edit Contacts* panel, which is described in Section 2.10 on page 30.
- **Later:** By-passes registration, taking you directly to the *Quick Look* panel. You can register at a later date. Refer to Section 18.2 on page 346.
- **Never:** Takes you directly to the *Quick Look* panel without ever registering the product.

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

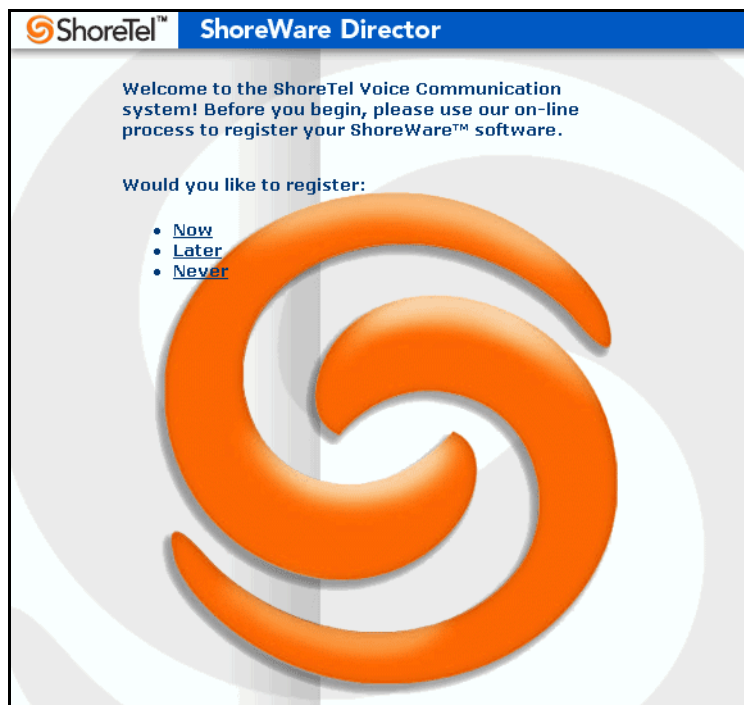


Figure 1-4 Product Registration panel

If your installation does not have adequate or current licenses, the *License Requirements* panel appears when you have completed or skipped registration. See Section 2.9 on page 25.

After you have registered ShoreWare Director, the *Quick Look* panel (shown in Figure 1-5) appears in the browser window.

Quick Look				Servers			Today's Events		
on localhost									
Switches									
Site	TMS Comm	Usage	Service	Server	Status	Services			
Headquarters	3/6	Idle	Lost Communications	Contact Center	In Service	Running	87	5	62
El Dorado	0/0			ShoreDoc-HQ	In Service	Running	24	82	101
Remote-Site	0/0								
Bonn	0/1	Unknown	Lost Communications						

Restart All Switches
Restart All Switches When Idle

© 1998-2004 ShoreTel, Inc. All rights reserved.

Figure 1-5 Quick Look panel

1.5 Navigation Frame

The navigation frame on the left (see Figure 1-6) provides access to the following:

- List, edit, and maintenance panels that appear in the data-entry frame on the right, as discussed in the next section
- Links to the system documentation
- A link for logging off from the system

It is more convenient to use the navigation frame links to navigate between panels appearing in the data-entry frame than to use your web browser navigation buttons.

Clicking the *Administration* link expands and contracts the administration links shown in Figure 1-6. Likewise, clicking the *Maintenance* or *Documentation* link expands and contracts their associated links.

Clicking a link further expands the list or takes you directly to a panel that appears in the data-entry frame. For example, clicking *System Parameters* expands the list; clicking *System Extensions* under the *System Parameters* link takes you to the *System Extensions* edit panel; and clicking *Switches* under the *System Parameters* link takes you to the *Switches* list panel.

1.5.1 Getting Help

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

- *ShoreTel 8 Administration Guide*
- *ShoreTel 8 Planning and Installation Guide*
- Telephone User Interface Guide IP Phone
- Telephone User Interface Guide Analog Phone
- Server Release Notes
- Client Release Notes

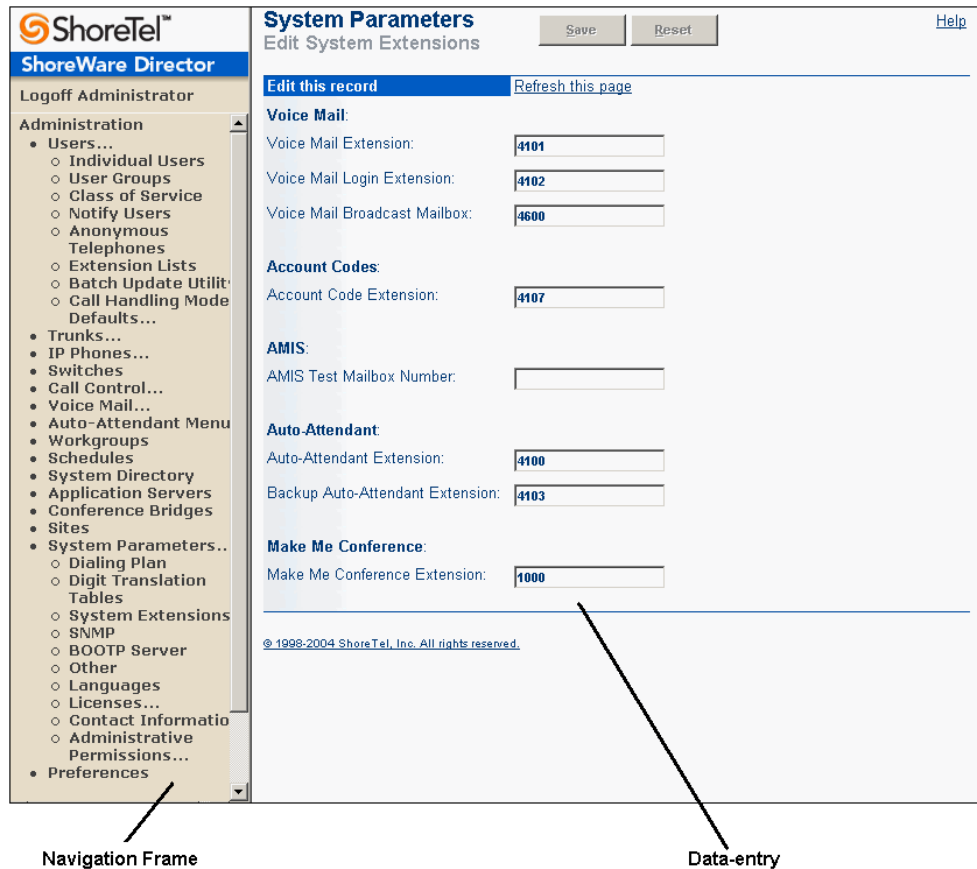


Figure 1-6 ShoreWare Director Layout

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

1.5.2 Logging Off

To log off from ShoreWare Director, click the *Logoff <user_name>* link in the navigation frame.

You are automatically logged off after 60 minutes of inactivity unless you are viewing the *Quick Look* panel or a *Switch* maintenance panel.

1.6 Data-Entry Frame

The data-entry frame provides a way to add, delete, or edit the system configuration, as well as to view status and issue commands to ShoreWare Director. These panels are referred to as list panels, edit panels, and maintenance panels.

1.6.1 List Panels

List panels provide tabular information about the current configuration and links for adding to or editing the configuration.

1.6.1.1 Adding

A *New* or *Add new* control at the top of most list panels allows you to make additions to the system configuration. Depending upon what is being added, you may have to select a site, trunk group, trunk type, switch type, or country. For example, the *Users* list panel shown in Figure 1-7 allows you to add a new user at a site by selecting the desired site and clicking *Go*. This brings you to the Individual Users edit panel to complete the configuration.

First Name	Last Name	Site	User Group	Client Type	Extension	Mailbox	Switch	Port	Status
Adam	Ant	Headquarters	<u>Executives</u>	Operator	4509	4509	Doc-24	ShoreDoc IP Phone	Home
Betty	Boop	Headquarters	<u>Executives</u>	Operator	4510	4510	Doc-24	10	Home
Carl	Castle	Headquarters	<u>Executives</u>	Operator	4511	4511	Doc-24	11	Home
Donald	Duck	Headquarters	<u>Executives</u>	Operator	4512	4512	ShoreDoc-HQ	Any IP Phone	Home
Ernie	Els	Headquarters	<u>Executives</u>	Personal	4513	4513	Doc-24	13	Home
Fred	Flintstone	Headquarters	<u>Executives</u>	Personal	4514	4514	Doc-24	14	Home
Greta	Garbo	Headquarters	<u>Executives</u>	Personal	4515	4515	Doc-24	15	Home
Howard	Hughes	Headquarters	<u>Executives</u>	Personal	4516	4516	Doc-24	16	Home
Johnathan	Seagull	Headquarters	<u>Executives</u>	Personal	4805	4805	Doc-24	1	Home
Tim	Toolman	Headquarters	<u>Executives</u>	Personal	4904	4904	Bedrock	10	Home
Ursula	Urp	Headquarters	<u>Executives</u>	Personal	4907	4907	Bedrock	13	Home
Vince	Vinceble	Headquarters	<u>Executives</u>	Personal	4906	4906	Bedrock	12	Home
Wanda	Wenderly	Headquarters	<u>Executives</u>	Personal	4908	4908	Bedrock	14	Home
Wendy	Williams	Headquarters	<u>Executives</u>	Personal	4905	4905	Bedrock	11	Home
Yourst	Yumbal	Headquarters	<u>Executives</u>	Personal	4903	4903	Bedrock	9	Home

Figure 1-7 Individual Users List panel

1.6.1.2 Linking to Edit Panels

List panels provide a way to navigate to individual records for the purpose of viewing, changing, or deleting information. For example, clicking a bold user name in the first column in Figure 1-7 takes you to the *Edit User* panel, where you can view and change the configuration of that user.

The items in bold on list panels link to the most common record to be changed. In addition, other items are underlined to allow fast navigation to related edit panels.

1.6.1.3 Sorting

The records on each list panel are presented in a default sort order and most panels allow you to change the sort order by clicking an associated column heading (ascending order only). For instance, clicking the *Extension column heading* on the *Individual Users* list panel sorts all users in ascending order by extension.

The column you click for sorting is the primary sort order. If you clicked a column other than the *First Name* column, the First Name column is the secondary sort order.

1.6.1.4 List Paging

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

1.6.2 Edit Panels

Edit panels include user interface controls that let you view and change individual system parameters. They also include command buttons that let you perform functions such as save and delete.

The screenshot shows the 'Edit User' interface. At the top, there are buttons for 'New', 'Copy', 'Save', 'Delete', and 'Reset'. Below these are navigation tabs: 'General' (selected), 'Personal Options', 'Distribution Lists', and 'Workgroups'. A 'Refresh this page' link is on the right. The form contains the following fields and options:

- First Name: Adam
- Last Name: Ant
- Number: 4509
- License Type: Extension and Mailbox
- Caller ID: (e.g. +1 (408) 331-3300)
- ✓ DID: +1213333 0001 (DID Range: +12133330000 - 0099)
- PSTN Failover: None
- User Group: Executives (with 'Go to this User Group' link)
- Site: Headquarters
- Language: English
- Home Port: IP Phones (selected), ShoreDoc IP Phone; Ports: Bedrock - 8; SoftSwitch: Contact Center
- Current Port: ShoreDoc IP Phone (with 'Go Home' button)
- Jack #:
- Mailbox on Server: ShoreDoc-HQ (with 'Voice Mail Delivery and Notifications' link)
- ✓ Accept Broadcast Messages
- ✓ Include in System Dial By Name Directory
- Make Number Private
- ✓ Enable Fax Redirection
- Enable Fax Server Integration
- ✓ Allow Use of Soft Phone
- Client Type: Operator
- Client User ID: AAnt
- Client Password: (masked)
- Voice Mail Password: (masked)
- Email Address: AAnt@yourcompany.com
- Conference Bridge:
 - Server: Collab
 - User ID: aant
 - Password: (masked)

At the bottom left, there is a link: [Edit System Directory Record](#).

Figure 1-8 Edit User panel

1.6.2.1 Control Buttons

This section describes the functions of the five control buttons that appear on most edit panels. Individual data-entry fields, drop-down lists, and option buttons are described in the appropriate sections throughout each chapter.

- **New:** Click to create a new record based on default values. This does not overwrite the values of the original record.

- **Copy:** Click to create a new record based on the relevant values of the current record.

This does not overwrite the values of the original record.

- **Save:** Click to save the record that you added or edited.
You receive a warning if you try to navigate to another panel before saving your changes.
- **Delete:** Click to delete the current record from the system.
- **Reset:** Click to reset the current record to the values previously saved.

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

1.7 Getting Started

This section lists a summary of tasks to perform to configure your system for the first time. This ordered list also follows the order in which this guide is organized. Before you begin configuring the system, make sure it has been properly installed as described in the *ShoreTel 8 Planning and Installation Guide*.

- ❑ Launch ShoreWare Director and log in using the default user ID (admin) and password (changeme).
- ❑ Register ShoreWare Director from the registration panel. You can by-pass registration using the *Never* option.
ShoreTel encourages you to register ShoreWare Director so that we have the most up-to-date information concerning your ShoreTel products and installation.
- ❑ If required, update your ShoreTel licenses from the *License Requirements* panel. If you have not previously updated your licenses, you will be directed to the *License Requirements* panel after you have registered ShoreWare Director.
Until you have updated all required licenses, ShoreWare Director will continue to open to the *License Requirements* panel after login.
- ❑ Configure system parameters from the *System Parameters* link:
 - Install any new licenses your installation requires from the *Licenses* panel.
 - Configure the system's dialing plan from the *Dialing Plan* panel. Review the extension length as well as the dialing plan reservations for extensions and trunk access codes.
 - Configure the system's extensions from the *System Extensions* panel. Review the default system extensions and change the defaults if you need to use these extensions for other purposes.
 - Review password and log file settings on the *Other* panel. You should be able to accept all the defaults.
 - Review *Administrative Permissions*. People who will have permission to change Director settings can be entered here.

After you have created users, you can return to this panel and give them administrative permissions.

You can typically by-pass the *Contact Information*, *SNMP*, and *BOOTP Server* panels.

- Configure sites by clicking the *Sites* navigation link:
 - Review the default Headquarters site information by clicking the *Headquarters* link on the *Site* list panel:
 - Country location for the site
 - Local area codes for 7- and 10-digit dialing
 - Emergency call back number (if using ISDN PRI)
 - Time zone (for correct date and time for caller ID telephones)
 - Admission control bandwidth (for multiple-site configurations)

You cannot configure *Night Bell Switch* or *Paging Extension* until the proper switch is configured, nor can you configure *Operator Extension* or *Fax Redirect Extension* until the proper users are configured. You will need to return to this panel later.
 - Set the IP address range for the IP phones at any remote sites. You define IP address ranges so that IP phones are assigned to the correct site. IP phones not assigned to a remote site are associated with Headquarters.
 - Configure additional sites if desired.
- Configure additional ShoreWare servers from the *Application Servers* panel:
 - Add each ShoreWare server.
 - Name each server and assign it to a site.
 - Set the voice mail and auto-attendant extensions.
 - Assign a user group to the server.
- Configure any ShoreTel Converged Conference Bridges in your system from the *Conference Bridges* panel. For more information, see the *ShoreWare Converged Conference Administration Guide*. Then configure the Conference Bridge server settings to enable document sharing and Make Me conferencing (see Section 11.4.1 on page 271).
- Configure the ShoreTel Contact Center Solution in your system. For more information, see the *ShoreTel Contact Center Solution Administration Guide* and the *ShoreTel Contact Center Planning and Installation Guide*.
- Configure the ShoreWare System Monitor. For more information, see the *ShoreWare System Monitor Guide*.
- Configure ShoreGear voice switches from the *Switches* panel:
 - Add each ShoreGear-120/24, ShoreGear 90, ShoreGear-60/12, ShoreGear 50, ShoreGear-40/8, ShoreGear 220T1/T1A/E1 and ShoreGear-T1/E1 voice switch.
 - Provide a name and description and use the *Find Switches* button to discover each voice switch on the network.
 - Assign each switch to the proper site.
 - Assign the configuration switches that manage the configuration downloads to the IP phones. If you do not assign configuration switches, the first two ShoreGear-120/24, ShoreGear-60/12, or ShoreGear-40/8 switches you configure are automatically set as configuration switches.

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

- Configure IP phones from the *IP Phones* link:
 - Add IP phone ports to ShoreGear-120/24, ShoreGear 90, ShoreGear-60/12, ShoreGear 50, ShoreGear-40/8, ShoreGear 220T1/T1A/E1 and ShoreGear-T1/E1 voice switches supporting IP phones. For each port you assign to IP phones, the switch supports five IP phones. For more information, see Chapter 8, starting on page 111.
 - Set the boot parameters of the IP phones. ShoreTel IP phones are set to find boot information from a DHCP server. If your installation has other requirements, use the IP phone set-up menu to set server and boot configuration parameters. For more information, see the *ShoreTel 8 Planning and Installation Guide*.

You can speed up the installation process by using the Office Anywhere feature. For more information, see Chapter 8, starting on page 111 as well as Appendix B, starting on page 399.

- Configure the following users before you add general users to the system:
 - During installation, a system administrator is set up. Assign a person at your site to this role. When you assign a system administrator, the default user ID and password must be changed. *Make a note of the new user ID and password, since the defaults (admin and changeme) will no longer be available.*
 - Configure an *Operator* for each site.

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

Configure a “user” as the *Fax Redirect* extension for each site.
 - Configure a user as the default *Personal Assistant* for all other users. This is the user that calling parties are routed to when they dial “0” in a user’s mailbox. It is important that you configure the default Personal Assistant before adding the bulk of the users so that appropriate defaults can be assigned. If you omit this step, you may have to spend time reconfiguring the users later.
 - Configure the *Call Handling Modes Defaults* and assign the *Personal Assistant*.
- Complete configuration of sites:
 - Return to each site and complete the configuration for *Night Bell*, *Paging*, *Operator*, and *Fax Redirect*.
 - If you have added additional servers, return to each Site edit panel and reconfigure as appropriate.
- Configure all trunk groups and trunks:
 - Configure trunk groups from the *Trunk Groups* panel. You can modify the default trunk groups, add new trunk groups, and assign individual trunks for Make Me conferencing.

- Configure the trunks from the *Individual Trunks* panel. For the ShoreGear-T1, return to the *Switch* edit panel and assign all the channels on the T1 to the proper trunk group.
- Configure the users from the *Users* panels:
 - Configure the user groups, including all the Class of Service (CoS) permissions from the *User Group* edit panel. You can modify the default user groups or add new trunk groups. Be sure to grant access to any new trunk groups you have added.
 - Configure all the users from the *Individual Users* edit panel.
 - Configure any anonymous telephones from the *Anonymous Telephone* edit panel.
- Configure call control parameters from the *Call Control* link. Set up hunt groups and paging groups, as needed.
- Configure voice mail parameters and system distribution lists from the *Voice Mail Options* and *System Distribution Lists* panels.
- Configure the auto-attendant parameters from the *Auto-Attendant* edit panels:
 - Configure each auto-attendant menu from the *Menus* panel.
- Set schedules from the *Schedules* link. These may be used by the auto-attendant or paging groups.
 - Configure the workgroups from the *Workgroups* edit panel:
 - Configure the workgroup.
 - Configure the call handling modes.
 - Configure the queue, including prompts.
- Configure the system directory from the *System Directory* edit panel:

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

Setting Up System Parameters

2.1 Overview

This chapter explains how to set up system-wide parameters from the *System Parameters* link. Click the *System Parameters* link in the navigation frame to display additional links, as shown in Figure 2-1.

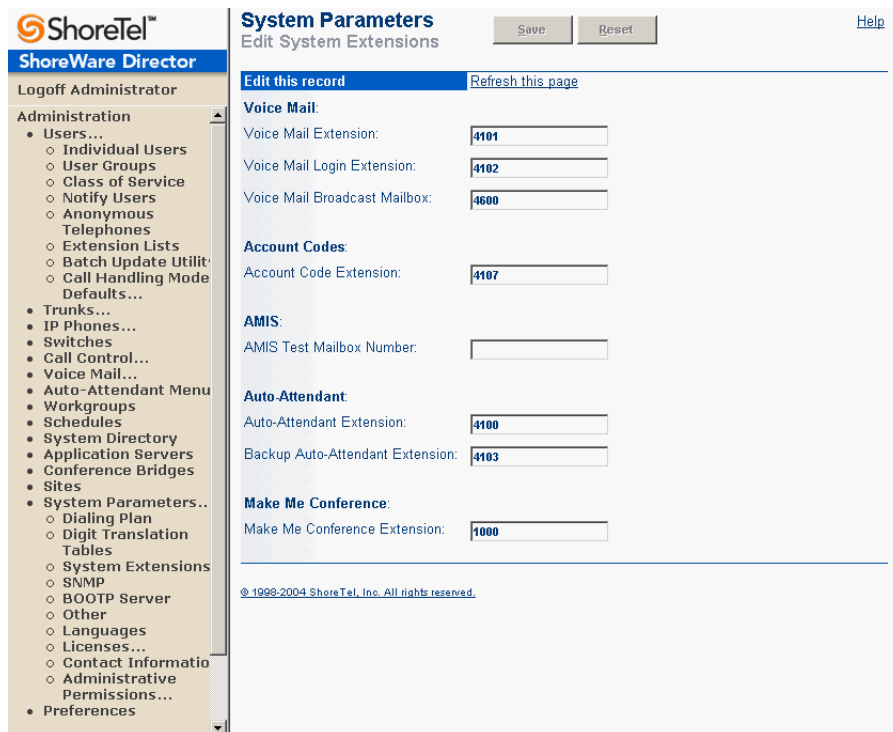


Figure 2-1 System Parameters Navigation Links

System parameters are grouped into categories corresponding to the sections in this chapter:

- Dialing Plan
- Digit Translation Tables
- System Extensions
- SNMP
- BOOTP Server
- Other Parameters
- Languages
- Licenses

- Contact Information
- Administrative Permissions

2.2 Dialing Plan

The *Edit Dialing Plan* page (Figure 2-2) lets you define the conventions for how extensions, trunk access codes, menu numbers, and distribution list numbers are managed by the system.

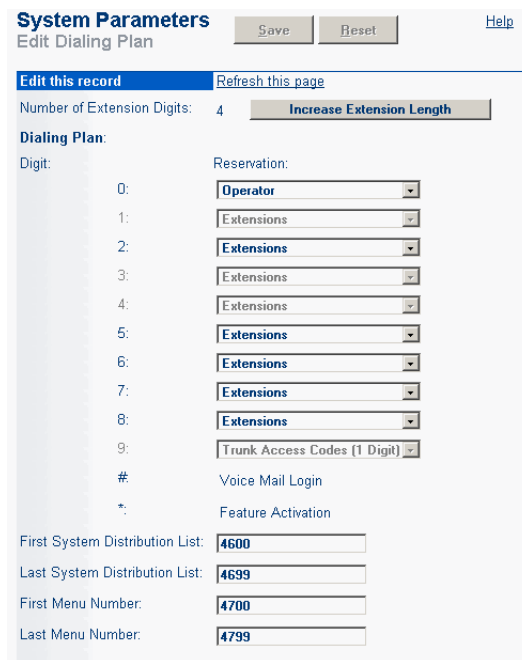


Figure 2-2 Edit Dialing Plan Page

To reach the Edit Dialing Plan page, click the *Dialing Plan* link under *System Parameters* in the navigation frame.

2.2.1 Parameters

The parameters on the *Edit Dialing Plan* page are as follows:

- **Number of Extension Digits:** This parameter tells you the current number of extension digits on the system. The ShoreTel 6.1 system supports extension lengths of 3, 4, or 5 digits. The default is 3 digits.

To increase the extension length, click the *Increase Extension Length* link. Due to the scope of the change, this procedure requires you to stop and start the ShoreWare voice services as well as the ShoreGear voice switches. Also see Section 2.2.2 on page 18.

- **Dialing Plan:** These parameters let you reserve digit ranges for the configuration of trunk access codes and dialed numbers.

The *Reservation* list lets you select one of the following:

- *Extension Prefix (n digits)* – For use with the On Net Dialing feature. 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.

Note: The Configure Extension Prefix Warning window appears with a list of each of the sites in your system. Next to the list of sites you will find a blank field that requires you to enter the desired extension prefix. This prefix will be applied to every dialed number at that particular site. Make sure to back up the system before clicking Save.

- **Extensions** – Reserves this range for use by extensions, mailboxes, distribution lists, and menus. Note that the digit “0” cannot be reserved for extensions.
- **Not Used** – Does not allow this range to be used.
- **Operator** – Defines the digit as a shortcut for reaching the site operator with the telephone. By default, the system is programmed with “0” as the operator digit.
- **Trunk Access Codes [1 Digit]** – Defines the range solely for 1-digit trunk access codes.
- **Trunk Access Codes [2 Digit]** – Defines the range solely for 2-digit trunk access codes.
- **Trunk Access Codes [3 Digit]** – Defines the range solely for 3-digit trunk access codes.

The ShoreTel 6.1 system provides a single, global image across sites and this needs to be considered in the dialing plan. Extensions, trunk access codes, operator digits, and emergency numbers for your various sites must not overlap.

- The operator digit defaults to “0”, but often the digit “0” is used as a trunk access code internationally. If you want to use it as a trunk access code, you will not be able to dial the operator with this digit. ShoreTel recommends that international customers standardize globally on a single trunk access code for the purposes of network call routing (for example, use “9” for all trunk groups).
- Extensions must not conflict with the leading digits of emergency telephone numbers, since the ShoreTel 6.1 system allows users to dial emergency numbers with or without a trunk access code. If you are deploying a global voice network, this must be considered for all emergency numbers. For instance, extensions should not include the following, since they relate to international emergency codes:

- * 112, 112x, 112xx (Parts of Europe)
- * 911, 911x, 911xx (U.S., Canada)
- * 999, 999x, 999xx (Parts of Europe and Asia/Pacific)

- **First System Distribution List and Last System Distribution List:** These parameters define the lower and upper boundaries, respectively, for the numbers reserved for system distribution lists. These values must be consistent with digit reservations for extensions.

When sending a voice mail message to a group of users who belong to a particular system distribution list, you enter the number of the distribution list. These lists are created under *System Distribution Lists* from the *Voice Mail* navigation link. See Section 12.1 on page 273 also.

- **First Menu Number and Last Menu Number:** These parameters define the lower and upper boundaries, respectively, for the numbers reserved for menu numbers. These values must be consistent with digit reservations for extensions.

The auto-attendant, in general, has an extension number defined on the **System Extensions** edit page. Menu numbers define the specific menu number on the system, since the system supports a total of 256 menus. See Section 13.2.2 on page 286 also.

2.2.2 Increasing Extension Length

You can increase the length of your phone extension digits. Due to the scope of this change, you are prompted to stop all ShoreWare voice services as well as restart the ShoreGear voice switches following the changes.

You can increase the extension length from:

- 3 to 4 digits
- 3 to 5 digits
- 4 to 5 digits

This is done by pre-pending one or two digits to all existing extensions, mailboxes, menus, and distribution lists.

To increase the extension length:

Step 1 Click *Increase Extension Length*.

A warning dialog appears as shown in Figure 2-3.

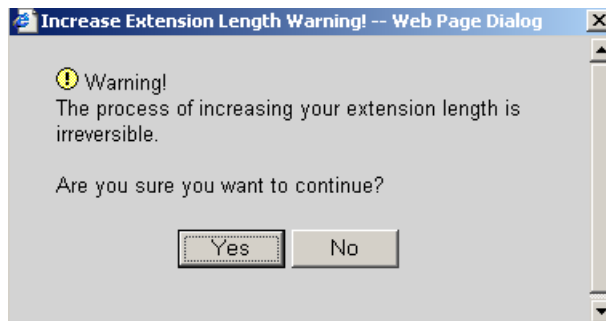


Figure 2-3 Warning Dialog Box

Step 2 Click *Yes* to increase the extension length.

Step 3 Specify the new digit or digits to be added in front of all existing and future extension numbers (see Figure 2-4). Click *OK* to apply the change.

2.3 Digit Translation Tables

Selecting *Digit Translation Tables* link from Parameters takes you to the Digit Translation List page shown in Figure 2-5. Select the list you wish to view or edit or click *New* to add a new table. An edit page, as shown in Figure 2-6 displays.

You can select an existing translation item for deletion by clicking the *Delete* button.

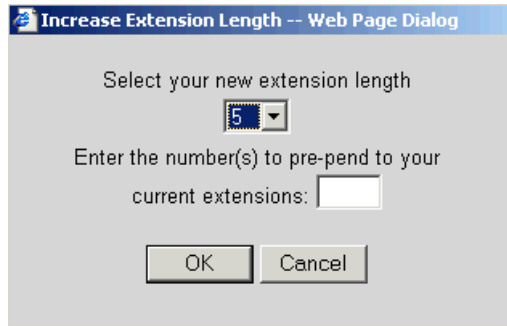


Figure 2-4 Specifying New Extension Digits

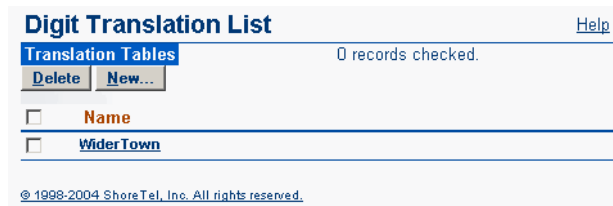


Figure 2-5 Digit Translation List Page

If you are using the application server as a PBX for a legacy voice mail system, you may have, for example, 4-digit extensions on the voice mail system and 3-digit extensions on the ShoreTel 6.1 system. To translate from one to the other, use the *Digit Translation Table*, as shown in Table 2-1. Also refer to Section 4.3 on page 47.

When using the application server as a PBX, set up the Digit Translation Table in the following way:

Actual Extensions		Digit Translation Table	
ShoreTel	Legacy VM	Original Digits	Replacement Digits
5xx	65xx	5	65
3xx	73xx	3	73
2xx	83xx	2	83
65xx	5xx	65	5
66xx	6xx	66	6
75xx	3xx	75	3

Table 2-1 Digit Translation Table

To create a digit translation table:

- Step 1** Click *New* and the Digit Translation page appears as shown in Figure 2-6.
- Step 2** Enter a name for the translation table and click *Save*.
- Step 3** Click *New* to add a translation rule in the Entry Dialog Box.
- Step 4** Click *Save* to save the rule.

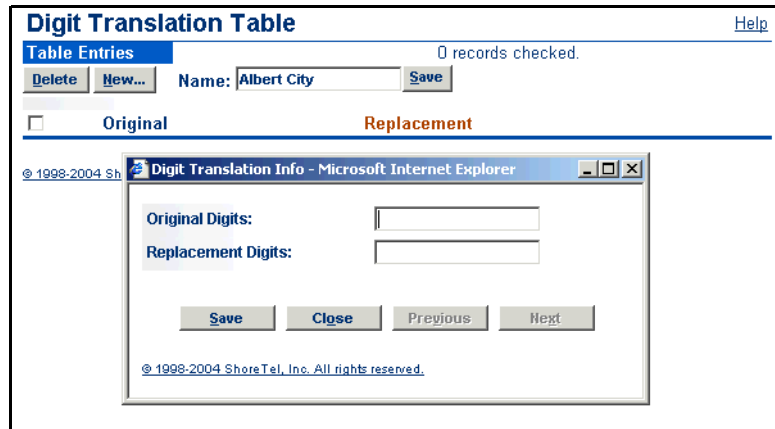


Figure 2-6 Digit Translation Page and Entry Dialog Box

2.4 System Extensions

The page is divided into five areas—*Voice Mail*, *Account Codes*, *AMIS*, *Auto-Attendant* and *Make Me Conference*—in which you configure the ShoreTel 6.1 system extensions. To invoke the *Systems Extensions* edit page (shown in Figure 2-7), click the *System Extensions* link under *System Parameters* in the navigation frame.

System Parameters	
Edit System Extensions	
<input type="button" value="Save"/> <input type="button" value="Reset"/>	
Edit this record	Refresh this page
Voice Mail:	
Voice Mail Extension:	<input type="text" value="5101"/>
Voice Mail Login Extension:	<input type="text" value="5102"/>
Voice Mail Broadcast Mailbox:	<input type="text" value="5600"/>
Account Codes:	
Account Code Extension:	<input type="text" value="5107"/>
Auto-Attendant:	
Auto-Attendant Extension:	<input type="text" value="5100"/>
Backup Auto-Attendant Extension:	<input type="text" value="5103"/>
Make Me Conference:	
Make Me Conference Extension:	<input type="text" value="5108"/>

Figure 2-7 System Extensions Edit panel

2.4.1 Parameters

The parameters on the *System Extensions* edit page are described below.

- **Voice Mail Extension:** This is the extension that is used for forwarding calls to voice mail. The value must be a valid, unused extension.

For example, if User A is configured to have calls forwarded to voice mail on ring no-answer, and User B calls User A, the call is routed to this extension after the no-answer number of rings.

Also see Chapter 12, starting on page 273.

- **Voice Mail Login Extension:** This is the extension that individual users call to log into their voice mailbox. For example, a user who wants to check voice messages can simply dial “#” (pound sign) or dial the voice mail login extension.

It is recommended that users be allowed to dial in from outside the company to retrieve their voice mail. Typically, you direct this number to an auto-attendant menu and configure the menu with a single-digit action of “Transfer to Extension” using the *Voice Mail Login Extension* parameter.

- **Voice Mail Broadcast Mailbox:** This is the mailbox number that is used for broadcasting a voice mail message to all users. It is similar to a system distribution list.
- **Account Code Extension:** This is the extension on the headquarters SoftSwitch associated with the account codes application. When account code collection is optional or forced, calls are routed to this extension for an account code prompt.

See also Section 9.2 on page 133.

- **Auto-Attendant Extension:** This is the extension associated with the auto-attendant. If you want to dial the auto-attendant, this is the extension you would use. Note that there is only one auto-attendant extension, but there can be multiple auto-attendant menu numbers.
- **Backup Auto-Attendant Extension:** This is the extension associated with the backup auto-attendant that is located in every ShoreGear voice switch.

The backup auto-attendant provides basic inbound call routing in case the auto-attendant on the ShoreWare server is unavailable. In addition, it answers calls routed to voice mail in case voice mail on the ShoreWare server is unavailable.

The backup auto-attendant is also used when extensions are not reachable during a network or switch outage and the Admissions Control Bandwidth is exceeded.

Callers who are accessing the ShoreTel 6.1 system over a SIP trunk can access to the Backup Auto-Attendant in the same manner as users who are accessing the system via all other trunk types. ShoreTel supports RFC2833 (DTMF), so if the voicemail server is down, external callers can enter an extension using DTMF to ring the extension of the user they are trying to reach.

- **Make Me Conference Extension:** This is the extension associated with the conference system. An available extension is configured at the time of installation and upgrade.

2.5 SNMP

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

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

Clicking the *SNMP* link under *System Parameters* in the navigation frame invokes the *SNMP* edit page (shown in Figure 2-8).

Figure 2-8 SNMP Edit panel

2.5.1 Parameters

The parameters that appear on the *SNMP edit* page are as follows:

- Community Strings:** This parameter lets you set the password for the Read-Only (Get) and Read/Write (Get/Set) strings.
 - When no community string is enabled, the SNMP agent is disabled.
 - *Read-Only (Get)*—Set this string to enable read-only SNMP access.
 - *Read/Write (Get/Set)*—Set this string to enable read-write SNMP access.
- Trap Receivers:** This parameter lets you define up to five trap receivers to receive traps from your ShoreGear voice switches. You must specify an IP address for each trap. You can use any IP address that has an SNMP trap listener (UDP Port 162) installed.

2.6 BOOTP Server

If your site uses a fixed IP address scheme, you can use the BOOTP server feature to assign IP addresses to voice switches. This is accomplished from the *BOOTP Server* edit page (Figure 2-9), which is invoked by clicking the *BOOTP Server* link from under *System Parameters* in the navigation frame. Refer to the *ShoreTel 8 Planning and Installation Guide* for more information on assigning IP addresses.

	Ethernet Address	IP Address	Subnet Mask	Default Gateway	
	00-10-49-00-00-10	10.0.0.2	255.255.255.0		Delete
Add this record	<input type="text" value="00-10-49-"/>	<input type="text"/>	<input type="text" value="255.255.255.0"/>	<input type="text"/>	

Figure 2-9 BOOTP Server Edit panel

The BOOTP server sends an IP address when requested by a ShoreGear voice switch whose MAC address is defined in the server's table. When setting up the BOOTP server's table, you enter each voice switch's MAC address and associate it with an IP address.

The subnet mask and default gateway addresses may change for ShoreGear voice switches on other subnets in WAN environments.

The columns in the table on the *BOOTP Server* edit page are as follows:

- **Ethernet Address:** This is a list of MAC addresses that are available for configuring your ShoreGear voice switches.
- **IP Address:** This is a list of IP addresses that are available for configuring your ShoreGear voice switches.
- **Subnet Mask:** This is your network's subnet mask.
- **Default Gateway:** This is typically set to the IP address of the nearest router that is configured to other configured switches and the ShoreWare server.

2.7 Other Parameters

A number of system-wide site parameters can be configured from the *Other Parameters* edit page (shown in Figure 2-10). To display this page, click the *Other* link under *System Parameters* in the navigation frame.

The screenshot shows the 'System Parameters' edit page. At the top, there are 'Save' and 'Reset' buttons, and a 'Help' link. Below the title, there is a 'Refresh this page' link and a '* modified' indicator. The page is divided into sections: 'Passwords', 'Log File Storage', and 'General'. Each section contains several input fields for configuration.

Section	Parameter	Value	Unit
Passwords:	Max Voice Mail Errors (2-50):	3	
	Min Voice Mail Password Length (4-26):	4	
	Min Client Password Length (4-26):	6	
	Max Client Password Length (4-26):	26	
Log File Storage:	Max Days (1-30):	7	
	Max Size (10-30000):	4000	MB
General:	<input checked="" type="checkbox"/> Prevent Users from Initiating Client Upgrades		

Figure 2-10 Other Parameters Edit panel

2.7.1 Parameters

The parameters grouped under **Passwords** are defined as follows:

- **Max Voice Mail Errors (2-50):** This parameter specifies how many times a caller can make an error when using the telephone to log into voice mail. When the user exceeds the number of errors specified by this parameter, the system informs the user of the error and hangs up.
- **Min Voice Mail Password Length (4-26):** This parameter specifies the minimum number of digits for the voice mail password needed to log into voice mail over the telephone.

- **Min Client Password Length and Max Client Password Length (4-26):** These parameters specify the minimum and maximum number of characters for the client password needed to log into the ShoreTel 6.1 Call Manager applications and ShoreWare Director.

The parameters grouped under *Log File Storage* are defined as follows:

- **Max Days (1–30):** This parameter specifies the number of days information will be stored in the log files on the ShoreWare server. Each evening, information older than the maximum days specified here is automatically deleted from the log files to help manage the amount of hard disk space used.
- **Max Size (10–30000) MB:** This parameter specifies the maximum size (in MB) of all log files on the ShoreWare server. This sets an upper bound on the amount of hard disk space used on the ShoreWare server for all log files.

The parameter under *General* is defined as follows:

- **Prevent Users from Initiating Client Upgrades (check box):** ShoreTel recommends selecting this check box to prevent the client machines in a ShoreTel 6.1 system from initiating an upgrade on their own. This should be selected if you will be using the Silent Client Upgrade feature (in tandem with Microsoft Active Directory) to install client software on remote machines. This approach allows an administrator to upgrade client software regardless of the permissions associated with the client machines (or the users logged into those machines).

2.8 Languages

ShoreTel 6.1 supports multiple languages. You can choose the languages that will be available in the system from the Language page, as shown in Figure 2-11, by checking the Enable box

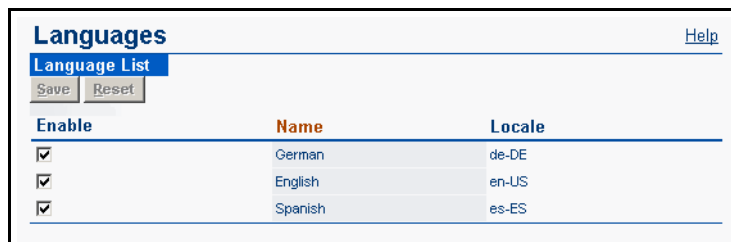


Figure 2-11 Languages List panel

If licensed, other languages can be selected from the following pages in Director:

- Site Edit
- Auto-Attendant Menu Edit
- User Edit
- Workgroup Edit
- Rout Point Edit
- Trunk Group Edit.

English is enabled by default, but it may be disabled and another language selected. If you try to disable a language already in use in the system, Director will display a warning before disabling the language.

2.9 Licenses

Clicking *Licenses* in the navigation frame shows two license page links: Requirements and Keys. The *License Requirements* page (shown in Figure 2-12) displays the keyed licenses and the self-audited licenses.

License Requirements Help		
License Requirement List		
Send/Print	Request System License Key	
Name	Required	Purchased
Keyed Licenses:		
ShoreWare System License (Enterprise Edition)	1	1
ShoreWare Additional Site License	5	10
ShoreWare Extension License	2	100
ShoreWare Mailbox License	2	100
ShoreWare SoftPhone License	0	10
ShoreWare Additional Language License	0	0
ShoreWare Remote Web Reporting License	0	0
ShoreWare Mobile Call Manager License	0	0
Self-audited Licenses:		
ShoreWare Personal Call Manager Software	0	0
ShoreWare Advanced Call Manager Software	2	22
ShoreWare Operator Call Manager Software	0	0
ShoreWare Workgroup Agent Call Manager Software	0	0
ShoreWare Workgroup Supervisor Call Manager Software	0	0
ShoreWare Remote Server Software	1	22
ShoreWare SIP Trunk License	2	22
ShoreWare TAPI Application Server	0	0
Shoreware Phone API License	0	0

Figure 2-12 License Requirements panel

The *License Requirements* page shows the licenses required for your system and allows you to track and manage all licenses. The licenses are divided into keyed and self-audited licenses.

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

Five Call Manger Licenses, each of which corresponds to a Call Manager type, span the complete Call Manager feature set. ShoreWare defines four of these licenses as Keyed Licenses.

2.9.1 Compliance

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

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

2.9.2 Node Locking – System key

Node locking ties a customer (and their ShoreTel software licenses) to a specific system. This is done using profile information on the main ShoreWare Director server. Customers and partners can submit their profile information online via ShoreWare Director. ShoreTel will then respond by sending an email with a new system key within 3-5 days.

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

To request a System Key, do the following:

- 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 **Licenses** link, followed by the **Keys** link.
- Step 4 Click the **Request System License Key** button.
- Step 5 Enter the appropriate values for the following fields:
 - Customer Name
 - Partner Name
 - Contact Name
 - Contact Email
 - Contact Phone
- Step 6 Click **Save to File** if you intend to print the request form and mail request to licensekeyrequest@shoretel.com. Otherwise, click the **Request Online** button to transfer the request to ShoreTel.
- Step 7 ShoreTel will send an acknowledgement, and within three business days you should receive the system and additional site license keys, if applicable.
- Step 8 Return to Director and click **Administration > System Parameters > Licenses > Keys** and click the **New** button to enter the license to the **Key** field.
- Step 9 Click **Save** to store your changes.

The warning message should now be cleared. If there is any license key not in compliance, the system license can not be applied.

2.9.3 Keyed License Types

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

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

- **ShoreWare System License:** This count includes licenses required on a per-system basis.

- **ShoreWare Additional Site License:** This count includes licenses required for each site beyond the main headquarter location. For installed base customers, when you upgrade and request your new system key, you will automatically receive additional site licenses for all configured sites.
- **ShoreWare Extension License:** This count includes all extensions licensed by both Extension Only and Extension and Mailbox licenses.
- **ShoreWare Mailbox License:** This count includes all mailboxes licensed by both Mailbox Only and Extension and Mailbox licenses.
- **ShoreWare SoftPhone License:** This count includes SoftPhone licenses, which are issued on a per-user basis. Obtain and install one licence for each SoftPhone user.
- **ShoreWare Additional Language Licenses:** This count includes licenses if more than one language is enabled.
- **Remote Web Reporting License:** This count includes one license if the ability to generate web-based CDR reports from a remote site is desired. Only one license is needed per system and this single license will allow many clients to use the feature.
- **ShoreWare Mobile Call Manager License:** This count includes licenses required for each client that is enabled for Mobile Call Manager.
- **ShoreWare Professional Call Manager License** is a keyed license provides access to the following:
 - All functions available through Personal Call Manager.
 - Instant Messaging Presence
 - Contact Viewer
- **ShoreWare Workgroup Agent Call Manager License** is keyed license that provides access to the following:
 - All functions available through Professional Call Manager.
 - Ability to transfer calls by dragging and dropping call cells into the buddy list.
 - Workgroup access utilities, including login, log out, and wrap up.
 - Workgroup Queue Monitor.
- **ShoreWare Workgroup Supervisor Call Manager License** is a keyed license provides access to the following:
 - All functions available through Workgroup Agent Call Manager.
 - Workgroup Agent Monitor.
- **ShoreWare Operator Call Manager License** is a keyed license that provides access to the following:
 - All functions available through Workgroup Supervisor Call Manager.
 - Advanced Buddy List functions, including pickup, unpark, edit buddies' active call properties, and edit call notes of buddies.
 - Bridge Call Appearance Monitor.
- **ShoreWare SIP Phone License** is a keyed license that enables the system to support one SIP device through a SIP proxy.
- **ShoreWare Standard Resolution Video License** is a keyed license that enables Call Manager to support one point to point video session at VGA resolution (640x480).
- **ShoreWare High Resolution Video License** is a keyed license that enables Call Manager to support one point to point video session at XGA resolution (1024x768).

2.9.4 Extension and Mailbox Licenses

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

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

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

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

Feature	Combo	Extension Only <i>Includes 3rd-party SMDI-based VM to ST PBX</i>	Mailbox Only <i>Includes ST SMDI-Based VM to 3rd-party PBX</i>
PBX features			
Use SoftPhone (requires SoftPhone license)	Yes	Yes	No
Make call, take call, etc.	Yes	Yes	No
Voice mail features			
Configure call handling modes	Yes	Yes	Yes
Forward calls to configured destination	Yes	Yes	No ^a
Create and play greetings	Yes	No	Yes
Use the Personal Assistant	Yes	No	Yes
Notification escalation	Yes	No	Yes
Configure Find Me	Yes	No	Yes
System call handling schedule	Yes	No	Yes
Create call handling notes	Yes	Yes	Yes
Assign Extension	Yes	No	Yes
Record name	Yes	No	Yes
Automated attendant features			
Dial by number, name	Yes	Yes	Yes ^b
Transfer to / Go to extension	Yes	Yes	Yes ^{Table b}
Message by number, name	Yes	Yes	Yes
Advanced features			
Office Anywhere	Yes	Yes	No
Member of a hunt group	Yes	Yes	No
Member of a workgroup	Yes	Yes	No
Call manager features			
Call Manager: Standard, Professional, Workgroup Agent, Workgroup Supervisor, Operator	Yes	No Mailbox Features	No extension features
Extension monitor	Yes	Operator only features	No
Agent monitor	Yes	No mailbox features	No

Table 2-2 Licensed Extension and Mailbox Feature Availability

Feature	Combo	Extension Only	Mailbox Only
		<i>Includes 3rd-party SMDI-based VM to ST PBX</i>	<i>Includes ST SMDI-Based VM to 3rd-party PBX</i>
Queue monitor	Yes	No mailbox features	No
Voicemail viewer	Yes	No	Yes
Call history	Yes	Yes	No
System directory	Yes	No mailbox features	No extension features
Outlook features			
Fwd voicemail as wav attachment	Yes	No	Yes
Voicemail form integration	Yes	No	Yes
Outlook Contact/Quick Dialer	Yes	Yes	No
Outlook Contact/Screen Pop	Yes	Yes	No
Outlook Calendar integration	Yes	Yes	Yes

Table 2-2 Licensed Extension and Mailbox Feature Availability

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

2.9.5 Keyed License Details

To view the details of a keyed license, click *Keys* under the *Licenses* link in the navigation frame. The *License Keys* page appears as shown in Figure 2-13. From the License Keys page you can add new licenses, delete old licenses, and e-mail or print license information.

License Key List			Help
			0 records checked.
<input type="button" value="Delete"/>	<input type="button" value="New..."/>	<input type="button" value="Send/Print"/>	
<input type="checkbox"/>	Key	Comment	Description
<input type="checkbox"/>	6DUL-VXWN-ONJD-3Q7Y:SPH-2	Two soft phones for Uplift	2 Soft Phone Licenses
<input type="checkbox"/>	Z4MR-H4QN-EQEU-KJN6:MSE-50	Key for 50 Multi-site users for the Documentation system	50 Multi-Site Enterprise Licenses

Figure 2-13 License Keys panel

To edit comments for a key, click the license key link. A dialog box with the key, comments, and description appears as shown in Figure 2-14. You can enter comments about the license in the comment text box.

2.9.6 Additional Language Licenses

ShoreTel 6.1 offers additional languages as a licensed option. Each available language is enabled from the Language edit page. For more informations, see Section 2.8 on page 24.

2.9.7 Self-Audited Licenses

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

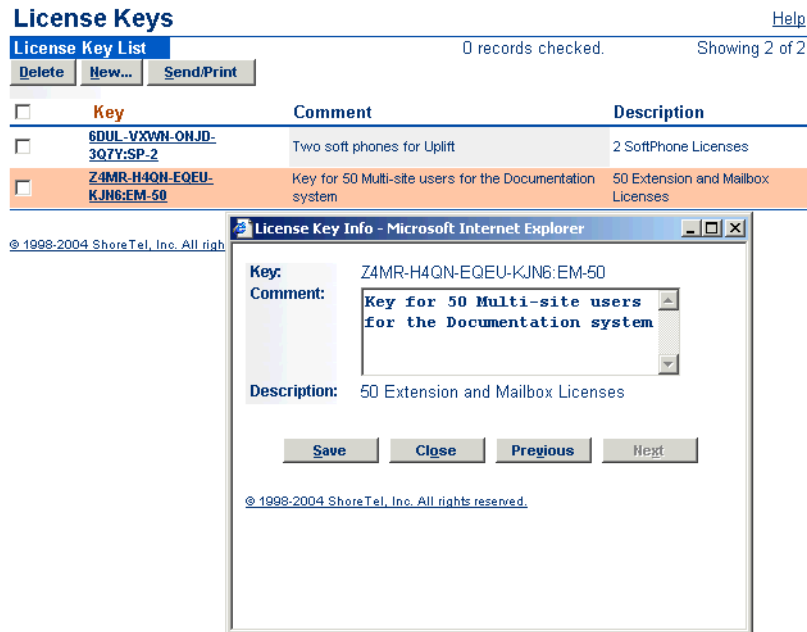


Figure 2-14 License Key Info dialog box

- **ShoreWare Personal Call Manager Software:** This count includes the number of ShoreWare Personal Call Manager licenses needed.
- **ShoreWare Remote Server Software:** This count includes licenses that correspond to additional ShoreWare servers, defined in Director, that correspond to additional voice mail servers. Up to 20 additional ShoreWare remote servers can be configured beyond the initial or headquarters server.
- **ShoreWare SIP Trunk:** This count includes licenses necessary for the implementation of SIP trunks.
- **TAPI Application Server:** This count includes licenses for remote TAPI Application Servers that have the “Allow Voice Mailboxes” check box deselected. The number purchased should match the number of deprioritized servers that exist at a particular site.
- **ShoreWare Phone API License:** This count includes licenses for the Phone API. (For more information, contact ShoreTel Professional Services for the appropriate SDK document.)

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

2.10 Contact Information

The *System Parameters Edit Contacts* page (shown in Figure 2-15) lets you provide contact information that is used for registering your ShoreTel system with ShoreTel, Inc. It includes the partner name, as well as the names and phone numbers of people who are administering and maintaining the system. This information is entered the first time ShoreWare Director is started, and is sent via email to ShoreTel, Inc.

System Parameters [Help](#)
 Edit Contacts

* modified

Edit this record [Refresh this page](#)

Registration Update:

Serial Number:

VAR ID Number:

Company:

Address:

City:

State/Province:

Postal Code:

Country:

Main Phone: (e.g. +1 (408) 331-3300)

Main E-mail:

Primary Contact

Name:

Title:

Phone: (e.g. +1 (408) 331-3300)

Pager:

E-mail:

Secondary Contact

Name:

Title:

Phone: (e.g. +1 (408) 331-3300)

Pager:

E-mail:

Figure 2-15 System Parameters Edit Contacts panel

2.10.1 Parameters

The parameters that appear on the *System Parameters Edit Contacts* page are as follows:

- **Registration Update:** Send the information online, or Print it and mail.
- **Partner Name:** This required field is the partner's name.
- **Customer Name:** This required field is the customer's name.
- **Address:** This required field is the customer's address.
- **City:** This required field is the customer's city.
- **State/Province:** This required field is the customer's state.
- **Postal Code:** This required field is the customer's zip code.
- **Country:** This required field is the customer's country.
- **Main Phone #:** This optional field is the customer's main telephone number.
- **Main E-mail:** This optional field is the customer's main e-mail address.

- **Primary Contact:** These required fields include information about the person or entity that administers and maintains the system. Information must be provided for the following fields:
 - Name
 - Title
 - Phone #
 - E-mail
- **Secondary Contact:** These fields provide information for a secondary contact and are optional.

2.10.2 Sending Contact Information

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

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

The MAC address for each ShoreGear voice switch will also be included in the registration e-mail.

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

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

2.11 Administrative Permissions

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

- Roles
- Administrators

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

Clicking the *New* button or an administrative title from the Role column invokes the *Edit Administrative Roles* page (see Figure 2-17). From here, you can define a new administrative role or change the permissions for an existing role.

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

Administrative Roles [Help](#)

Administrative Role List 0 records checked.

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

Figure 2-16 The Administrative Roles List panel

2.11.1 Parameters

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

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

Administrative Roles

Edit Administrative Role

Edit this record [Refresh this page](#)

Name:

Administrative Permissions Management

Account Code Management

System Directory Management

Report Generation Management

All Other System Management

User Management:

None

All Sites

Selected Sites

Aca-dack HeadQuarters Ireland test site Karoten Killah_site Remote_Site	<input type="button" value="Add >>"/>	<input type="button" value="Remove <<"/>
--	---	--

User Group and License Type Assignment:

None

All User Groups

Selected User Groups

Account Code Service Anonymous Telephones Copy of Executives Executives Executives Test UG House Telephones	<input type="button" value="Add >>"/>	<input type="button" value="Remove <<"/>
--	---	--

Distribution List Membership Assignment:

None

All Distribution Lists

Selected Distribution Lists

Broadcast Executives Managers SDL Staff	<input type="button" value="Add >>"/>	<input type="button" value="Remove <<"/>
--	---	--

Basic Workgroup Management:

None

All Workgroups

Selected Workgroups

dumas' Workgroup Hedgehog HQ Workgroup 8-58 Interflow WG IP230 Workgroup Remote WG (3-5)	<input type="button" value="Add >>"/>	<input type="button" value="Remove <<"/>
---	---	--

Site (Switches, Trunks, IP Phones, Servers) Management:

None

All Sites

Selected Sites

Aca-dack HeadQuarters	<input type="button" value="Add >>"/>	<input type="button" value="Remove <<"/>
--------------------------	---	--

Figure 2-17 Adding or Editing Administrative Roles

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

Deny permission by clicking None.

- **User Group Assignment:** Permission to add users to or move users between user groups may be granted for all sites or for a set of selected sites. Click *All User Groups* to grant permission system-wide. Click *Selected User Groups* to limit permissions, then highlight the user groups to be permitted, and click *Add* to move them to the permitted list. Be sure to select all groups you may be moving users to or from.

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

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

Deny permission by checking None.

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

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

Deny permission by checking None.

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

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

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

Deny permission by checking None.

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

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

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

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

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

ShoreWare Director is delivered with the following default Administrative Roles:

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

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

Administrative Roles [Help](#)

Administrative Role List 0 records checked.

[Delete](#) [New...](#)

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

Figure 2-18 The Administrative Roles List Page

2.11.2 Assigning an Administrative Role to a User

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

Administrators [Help](#)

Administrator List 0 records checked.

[Delete](#) [New...](#)

<input type="checkbox"/>	First Name	Last Name	User ID	Administrative Role
<input type="checkbox"/>	Fred	Flintstone	FFlintstone	Technical Support
<input type="checkbox"/>	Greta	Garbo	GGarbo	Technical Support

Figure 2-19 The Administrator List Page

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

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

To assign an administrative role to a user, click *New* from the *Administrators List* page. The pop-up shown in Figure 2-20 lets you type a *User* name or click *Search* to select from a list of users. Then assign a *Role* from the drop-down list.

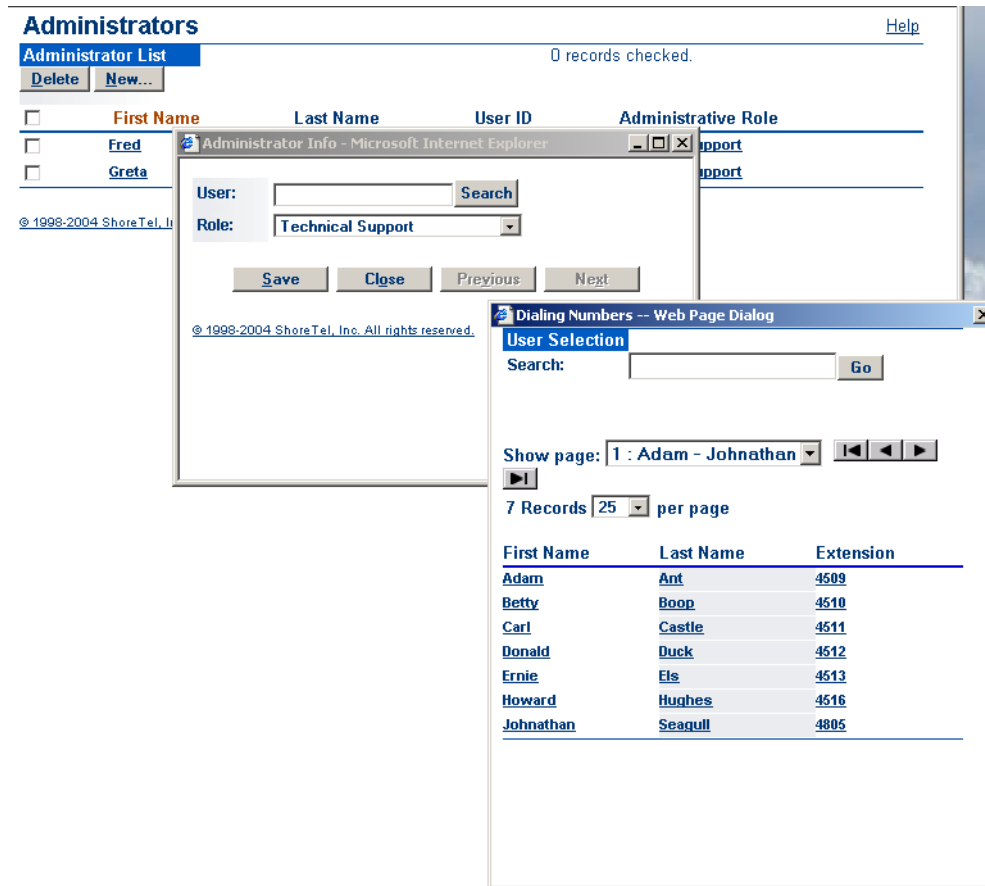


Figure 2-20 Assigning an Administrative Role to a User

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

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

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

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

ShoreTel Sites

This chapter explains how sites are implemented and configured in a ShoreTel 6.1 system. This chapter provides information on the:

- Site concept
- Sites list page
- Site edit page and parameters

3.1 Overview

ShoreTel 6.1 supports the concept of sites. A site is defined as a geographical location that has ShoreGear voice switches. This can be a headquarters site or a regional office.

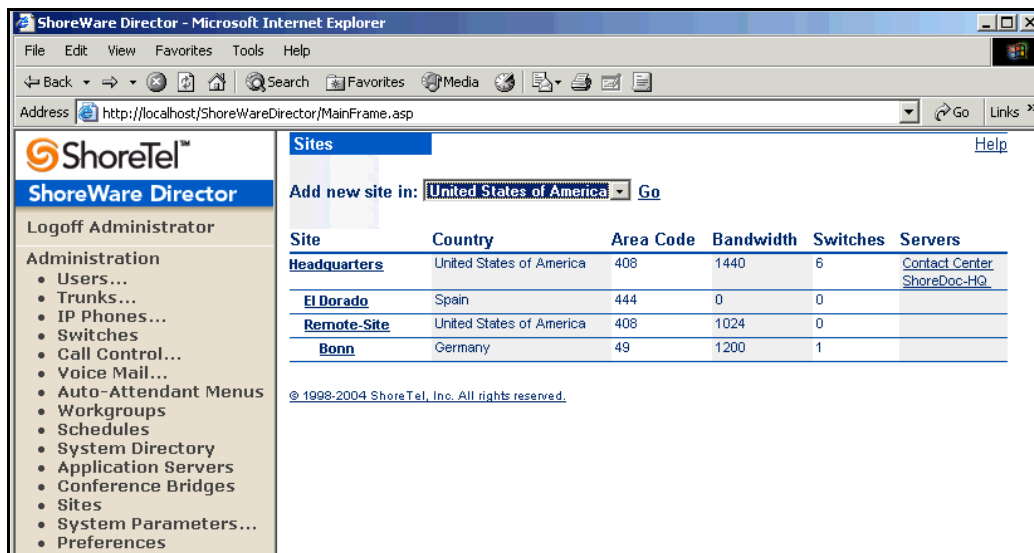


Figure 3-1 Sites List Page

Configuring sites lets you assign features to the associated site. For example, a site has a country, a local area code, and a site operator, as well as an admission control setting.

ShoreTel 6.1, by default, is configured with one site, called “Headquarters.”

3.2 Configuring Sites

The Sites link in the ShoreWare Director navigation frame gives you access to the *Sites* list page as shown in Figure 3-1.

The *Sites* list page presents sites organized in a hierarchical tree beginning with the Headquarters site. To add a new site, select a country and click *Go*. The *Sites* edit page appears, as shown in Figure 3-2. To edit an existing site, click the site name listed under the *Sites* column on the *Sites* list page.

Figure 3-2 Sites Edit Page

3.3 Parameters

The parameters on the *Sites* edit page are as follows:

- **Name:** This is the name of a new or existing site. It must be unique.
- **Country:** This is the name of the country in which the site is located.
- **Language:** This is the default language for the site. You must obtain a license to enable more than one language. For more information see Section 2.9 on page 25.
- **Parent:** The default parent site is Headquarters. Headquarters does not display the drop-down list of sites. Sites other than Headquarters must select a parent. This server is used for two purposes:
 - *By ShoreWare Director* to provide a default server when new users are added
 - *By the call control software* in the ShoreGear voice switches so that it knows where to route calls that request voice mail service

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

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

The Extension Prefix field will not appear in this window until after you have modified the Dialing Plan window, thus enabling the On Net Dialing feature.

- **Local Area Code:** This defines the local area code of the site so that users can dial local numbers without an area code. In the United States, this is the area code used for 7-digit dialing. For example, when the user dials an access code followed by seven digits at the site, this is the area code they are dialing.

This also defines the area code that is considered “local” from a call permissions point of view.

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

This also defines additional area codes that are considered “local” from a call permissions point of view.

- **Caller’s Emergency Service Identification (CESID):** The Caller’s Emergency Service ID (CESID) is the telephone number sent to the service provider when a user dials an emergency services number (e.g., 911 in the U.S.). This feature is only applicable to T1 PRI trunks. See Appendix B, starting on page 399 for more information.
- **Time Zone:** This is the site’s time zone that is associated with the ShoreGear switches and is used to deliver the correct time and date to Caller ID telephones.
- **Night Bell Extension:** This is the extension that is used to ring the site’s night bell. This extension must be associated with a ShoreGear switch audio output port that you specify as the next parameter. This extension is unique.

You must configure the appropriate switch before assigning the night bell extension.

- **Night Bell Switch:** This is the ShoreGear switch associated with the night bell extension. The night bell extension can share the same switch port as the paging extension.
- **Paging Extension:** This is the extension used for your overhead paging system. This extension must be associated with a ShoreGear switch audio output port that you specify as the next parameter. There is only one paging extension per site.
You must assign switches to the site and select the switch that will support the paging extension before you can save a paging extension.
- **Paging Switch:** This is the ShoreGear switch associated with the paging extension. The paging extension can share the same switch port as the night bell extension.
- **Operator Extension:** This is the extension to which the user is transferred when dialing the operator digit for the site (typically “0”).

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

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

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

The fax redirection extension must be an existing user.

- **Admission Control Bandwidth:** This defines the bandwidth that voice media streams can consume between this site and all other sites. The caller hears a "network busy" prompt if this value is exceeded. To compute the admission control value for the site, see Chapter 9, "Network Requirements and Preparation" in the *ShoreTel 8 Planning and Installation Guide*.
- **Emergency Number List:** This is the list of numbers that can be dialed at the site with or *without* a trunk access code for emergency services. Note that this number must not conflict with any extensions.
 - **Trunk Access Code Required** – When this checkbox is selected, a caller must dial the Trunk Access Code before dialing the specified emergency number. If not selected, entering the Trunk Access Code before the Emergency number is permitted, but not required, to complete the call.
 - **Data Entry Field** – Enter the exact emergency number required to contact the associated Emergency Service Provider. If Trunk Access Code Required is selected, you can also enter a number in canonical format.
 - **Add More** – Click this button to create additional *Data Entry Fields* for entering additional emergency numbers. Each site is permitted to have a maximum of ten emergency numbers to accommodate locations where multiple emergency service numbers are required.
- **Edit IP Phone Address Map:** This link opens the *IP Phone Address Map Info* edit page where you set the IP address range for sites other than Headquarters (see Figure 3-3). All IP phones are assigned to Headquarters by default. If Headquarters is your only site, you do not need to set IP address ranges. If you have more than one site with IP phones, you must set an IP address range for each site (other than Headquarters).

This page is also accessible from the IP Phone Address edit page. For more information, see Chapter 8, starting on page 111.

- **Site:** If you are setting the IP address range for a site other than the one shown in the *Site* drop-down list, select it from the list.
- **Low IP Address:** This is the lowest IP address of the range of addresses.

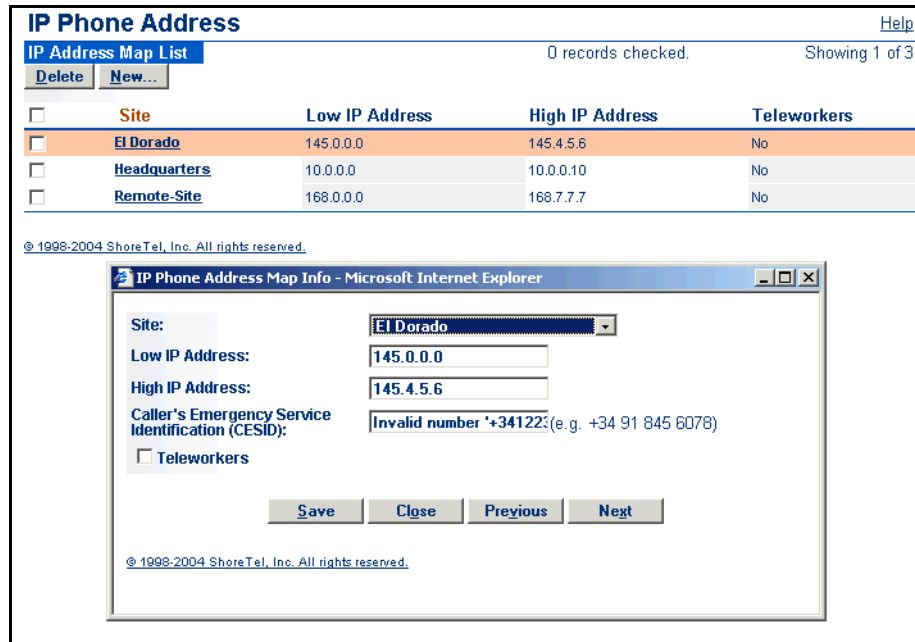


Figure 3-3 IP Phone Address Map Edit Page

- **High IP Address:** This is the highest IP address of the range of addresses.
- **Caller's Emergency Service Identification (CESID):** Enter the Caller's Emergency Service ID to be used for IP phones in this IP address range. Enter, for example, +1 (408) 555-5555. This is the telephone number sent to the service provider when a user dials an emergency services number (e.g., 911 in the U.S.) and does not have a DID number or is in a user group for which the DID number is not to be used as CESID. This feature is only applicable to ISDN PRI trunks.
- **Teleworkers:** This call will be considered an intersite call using intersite codecs and will adjust the bandwidth at the receiving site.

Configuring Application Servers

This chapter explains how to set up servers in a ShoreTel 6.1 system. This chapter provides information on:

- ShoreTel servers
- Distributed voice mail servers
- Legacy voice mail integration
- How to add servers to the system

4.1 Overview

The ShoreTel 6.1 system supports not only distributed call control, but also distributed voice application servers. Distributed servers are extremely valuable for two purposes:

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

Even though there are multiple servers, the ShoreTel 6.1 system provides a single image of your entire network. The system is currently certified to support up to 21 servers, one main server and up to 20 distributed servers. You should begin to consider adding a server at a site when the site exceeds 100 users. You also need to add a new server for every 1,000 users.

The distributed servers run the following voice applications:

- **Voice Mail** – Each server supports 254 simultaneous voice mail or auto-attendant connections. The voice mail system uses SMTP to transport composed messages between the distributed servers. The ShoreTel 6.1 system also supports linking to legacy voice mail systems using AMIS protocols.
- **Auto-Attendant** – The system supports up to 256 menus that are hosted on every server, and each server provides 64 voice mail/auto-attendant connections.
- **Configuration** – The system enables users to log in and make configuration changes (call handling modes, etc.) from their ShoreTel 6.1 Call Managers or from the Web Access call handling mode client.
- **Maintenance** – The system provides a web site accessible through ShoreWare Director for maintenance of all the remote servers.

The distributed voice applications use a Remote TAPI Service Provider that relies on the call control information from the main server. When you want to add reliability to your remote server, consider using redundant network paths to the main server.

4.2 Distributed Voice Mail

The ShoreTel 6.1 system uses Distributed Voice Mail (DVM) to provide greater voice mail availability. Each ShoreWare remote server has an instance of the telephony platform included, allowing voice mail and auto-attendant services to maintain full functionality during WAN short-term outages. The enhanced DVM included with the ShoreWare remote server allows users with mailboxes on the remote server to receive and pickup voice mail messages without depending on WAN connectivity to the headquarters server. The message waiting indicator (MWI) lights will correctly update with or without WAN connectivity.

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

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

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

4.2.1 IP Phone Limitations/Requirements

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

- **Directory feature:** In addition to basic connectivity, the directory feature also requires connectivity between the switch and a headquarters (HQ) server or distributed voice mail (DVM) server that controls that switch.
- **Options features, Changes to Call Handling Mode (CHM), Wrap-Up:** In addition to basic connectivity, these features require connectivity between the switch and an HQ server or DVM server that controls that switch. In addition, if the aforementioned switch is a DVM server, connectivity is required between that server and the HQ server. Further, connectivity between the DVM server and the HQ server is required for successful writes to the database.
- **Switch-to-switch extension monitoring:** This condition exists when a programmed button requires monitoring activity on an extension that is serviced by a different switch than the one that controls the phone. In this instance, call the user's switch A, and the monitored extension's switch B. Switch A is controlled by server X, switch B is controlled by server Y. Servers X and Y may each be DVM or HQ servers, and they may be the same server or different servers. For proper functionality of the switch-to-switch extension monitoring, the following conditions must exist:
 - Switch A must be able to talk to server X.
 - Server X must be able to talk to server Y.

- Server Y must be able to talk to switch B.
- If X and Y are the same, connectivity is, of course, assumed to exist.
- Auxiliary information on incoming calls, such as trunk information and called workgroup (WG) information, requires connectivity between the switch and a headquarters (HQ) server or distributed voice mail (DVM) server that controls that switch.

4.2.2 Call Manager Limitations/Requirements

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

4.3 Legacy Voice Mail Integration

ShoreTel 6.1 seamlessly integrates with legacy phone systems for customers who would like to have the freedom and flexibility to continue to use their legacy systems while migrating toward a newer IP telephony solution. The legacy system must continue to work flawlessly regardless of whether calls are traversing the ShoreTel 6.1 PBX on their way to the legacy voice mail system or whether they are traversing the legacy PBX on their way to ShoreTel 6.1 voice mail.

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

There are two modes of operation with respect to integrating a ShoreTel 6.1 system and a legacy system using SMDI:

- **External voice mail** – In this configuration, the legacy system provides voice mail services while the ShoreTel 6.1 system acts as the PBX for users.
- **ShoreTel voice mail** – In this configuration, the ShoreTel 6.1 system provides voice mail services while the legacy system acts as a the PBX for users.

Voice mail extension lengths for the legacy voice mail system may be different from the ShoreTel voice mail extension lengths. In this case, digit translation information is required. For more information on digit translation tables, see Section 2.3 on page 18.

For more information about integration to legacy voice mail systems, see the *ShoreTel 8 Planning and Installation Guide*.

4.4 Adding Application Servers

The *Application Servers* link in the ShoreWare Director navigation frame gives you access to the *Application Servers* list page as shown in Figure 4-1.

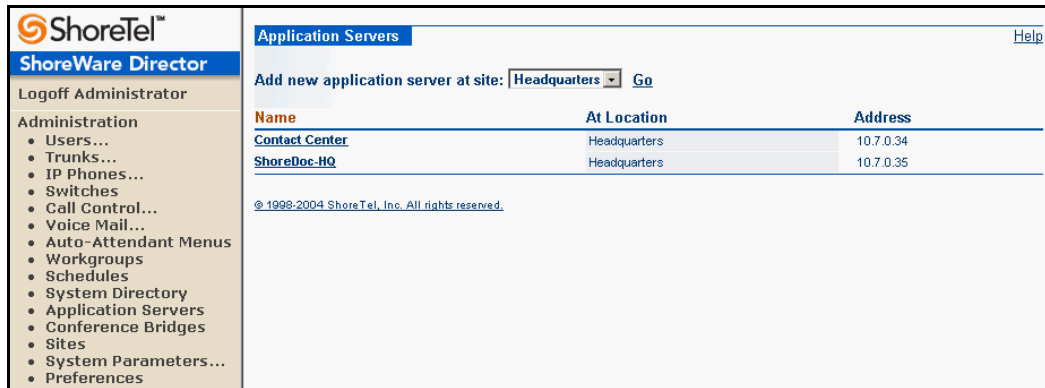


Figure 4-1 Application Servers List Page

The *Application Servers* list page is presented in alphabetical order.

The ShoreTel 6.1 system, by default, is configured with one server at the “Headquarters” site. Additional sites must have been configured prior to this step to be available. Refer to Chapter 3, starting on page 39 for more information about adding remote sites.

To add a new server:

- Step 1 Select the server site from the *Add new application server at site* drop-down list.
- Step 2 Click *Go*. The *Application Servers* edit page appears as shown in Figure 4-2.
- Step 3 Enter parameters for the new server as described below.
- Step 4 Click *Save*.

To edit an existing server:

- Step 1 Select a server from the list in the *Name* column by clicking on it.
- Step 2 The *Application Servers* edit page appears, as shown in Figure 4-2.
- Step 3 Change parameters as needed for the new server.
- Step 4 Click *Save*.

4.4.1 Parameters

The parameters on the *Application Servers* edit page are as follows:

- **Name:** This is the name of a new or existing server.
- **Host IP Address:** This is the IP address of the server.

If you enter the server host name, Director resolves the IP address when you click **Ping this Server**.

Application Servers
Edit Server

New Copy Save Delete Reset

Edit this record Refresh this page

Name: SoreDoc-HQ

Host IP Address: 10.1.2.3 Ping this Server

Site: HeadQuarters

SoftSwitch Name: SoreDoc-HQ

Maximum Trunks for Voice Mail Notification (1 - 200): 4

Allow Voice Mailboxes

Voice Mail and Auto Attendant:

Voice Mail Extension: 555-8885

Voice Mail Login Extension: 555-8886

Auto Attendant Extension: 555-8887

Assigned User Group: Voice Mail Notification

Default Auto-Attendant Menu: AA Menu1

Simplified Message Desk Interface:

Mode: External Voice Mail

COM Port (1 - 10): 2

Message Desk Number (0 - 999): 22

Number of Digits (2 - 32): 10

Translation Table: test Edit this translation table

Use for Call Data

Use for MWI Data

Extension List (extension - port - logical terminal number):

Add Edit Remove

Figure 4-2 Application Servers Edit Page, SMDI for External Voice Mail

You can also use the **Ping this Server** button to test the connectivity between your client PC and the new server.

- **Site:** This is the physical location of the server. The location of the server is used to calculate bandwidth consumption for the purposes of admission control.
- **SoftSwitch Name:** This is the name of the SoftSwitch on the server you are editing. ShoreTel 6.1 automatically creates a SoftSwitch for each server on the system.
- **Maximum Trunks for Voice Mail Notification (1 - 200):** This is the maximum number of trunks that can be used in the event of a voice mail notification. If many escalation profiles have been configured, it may be desirable to set this to a relatively low number to prevent notifications from overwhelming the system and making it impossible for users to make an outbound call.

Application Servers

Edit Server

New Copy Save Delete Reset

Edit this record
Refresh this page

Name:	<input type="text" value="Hq Server"/>	
Host IP Address:	<input type="text" value="10.11.11.100"/>	Ping this Server
Site:	HeadQuarters	
SoftSwitch Name:	<input type="text" value="HQ_softSwitch"/>	
Maximum Trunks for Voice Mail Notification (1 - 200):	<input type="text" value="4"/>	
<input checked="" type="checkbox"/> Allow Voice Mailboxes		
Voice Mail and Auto Attendant:		
Voice Mail Extension:	<input type="text" value="888-5511"/>	
Voice Mail Login Extension:	<input type="text" value="888-5512"/>	
Auto Attendant Extension:	<input type="text" value="888-5513"/>	
Assigned User Group:	<input type="text" value="Voice Mail Notification"/>	
Default Auto-Attendant Menu:	<input type="text" value="Default AA"/>	
Simplified Message Desk Interface:		
Mode:	<input type="text" value="ShoreTel Voice Mail"/>	
Trunk Group:	<input type="text" value="SMDI"/>	
COM Port (1 - 10):	<input type="text" value="1"/>	
Message Desk Number (0 - 999):	<input type="text" value="1"/>	
Number of Digits (2 - 32):	<input type="text" value="7"/>	
Translation Table:	<input type="text" value="<None>"/>	
<input type="checkbox"/> Use for Call Data		
<input type="checkbox"/> Use for MWI Data		
<input checked="" type="checkbox"/> Use Flash to Route Calls		

Figure 4-3 Application Servers Edit Page, SMDI for ShoreTel Voice Mail

- **Allow Voice Mailboxes:** Select this checkbox to allow voice mailboxes on this server. Clear this checkbox to prevent voice mailboxes to exist on this server.

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

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

By default, this checkbox is selected.

Voice Mail and Auto-Attendant

- **Voice Mail Extension:** This extension is used by the voice mail server.
- **Voice Mail Login Extension:** This extension is used to log in to the voice mail server.
- **Auto-Attendant Extension:** This extension is used by the auto-attendant server.
- **Assigned User Group:** This is the assigned user group for the server. Because voice mail places outbound calls, the server must have assigned permissions.

- **Default Auto-Attendant Menu:** Each server can have a different default auto-attendant menu. This is the menu reached when none is specified—for instance, when a caller dials “9” to escape from voice mail and return to the auto-attendant.

Simplified Message Desk Interface (SMDI) External Voice Mail

- **External Voice Mail:** If this application server is to function as a PBX for a legacy voice mail system, check this box.
- **COM Port (1-10):** This is the COM port used by SMDI.
- **Message Desk Number (1-999):** The Message Desk default is 1. Valid values are 1 through 999. Set the number that the voice mail system expects. This parameter is most often set to one, since only one system will be using the SMDI link. In some configurations, however, a number of SMDI links can be daisy-chained together and the Message Desk Number value is used to allow each system to know which data belongs to it.
- **Number of Digits (2-32):** This field sets the number of digits the ShoreTel 6.1 system sends in the SMDI extension fields. Set this number to the value the voice mail system expects, most commonly 7 or 10. If the number of digits and the ShoreTel 6.1 system extension value differ, the extension number is padded. For example, if ShoreTel needs to send extension 456 and the Number of Digits field is equal to 7, extension 0000456 is sent. If no padding is desired, the Number of Digits field would be set to 2 in this example. Then, only 456 is sent.
- **Translation Table Use for Call Data:** This check box indicates that the digit translation table is to be used for call data, when checked. Both Translation Table boxes may be checked at the same time.
- **Translation Table Use for MWI Data:** This check box indicates that the digit translation table is to be used for Message Waiting Indicator data, when checked. Both Translation Table boxes may be checked at the same time.
- **Extension List (extension - port - logical terminal number):** The SMDI message must contain the user extension, port number, and logical terminal number (exact trunk number). Note that these extensions forward to the Backup Auto-Attendant on No Answer or Busy.

Simplified Message Desk Interface (SMDI) ShoreTel Voice Mail

- **ShoreTel Voice Mail:** If this application server is to function as a voice mail server for a legacy PBX, check this box.
- **Trunk Group:** Select the trunk group to be used by the legacy PBX for voice mail traffic.
- **COM Port (1-10):** This is the COM port used by SMDI.
- **Message Desk Number (1-999):** The Message Desk default is 1. Valid values are 1 through 999. Set the number that the voice mail system expects. This parameter is most often set to one, since only one system will be using the SMDI link. In some configurations, however, a number of SMDI links can be daisy-chained together and the Message Desk Number value is used to allow each system to know which data belongs to it.
- **Number of Digits (2-32):** This field sets the number of digits the ShoreTel 6.1 system sends in the SMDI extension fields. Set this number to the value the voice mail system expects, most commonly 7 or 10. If the number of digits and the ShoreTel 6.1 system extension value differ, the extension number is padded. For

example, if ShoreTel needs to send extension 456 and the Number of Digits field is equal to 7, extension 0000456 is sent. If no padding is desired, the Number of Digits field would be set to 2 in this example. Then, only 456 is sent.

- **Translation Table:** Select a translation table from the drop-down list. For information on creating translation tables, see Section 2.3 on page 18.
- **Use for Call Data:** This check box indicates that the digit translation table is to be used for call data, when checked. Both Translation Table boxes may be checked at the same time.
- **Use for MWI Data:** This check box indicates that the digit translation table is to be used for Message Waiting Indicator data, when checked. Both Translation Table boxes may be checked at the same time.
- **Use Flash to Route Calls:** Select this checkbox to use flash (i.e. a short hang-up to provide signaling instructions to a PBX) to route calls between the ShoreTel voicemail system and the legacy PBX. Enabling this feature may result in a more efficient trunk allocation.

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

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

4.4.2 Extension List Mapping

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

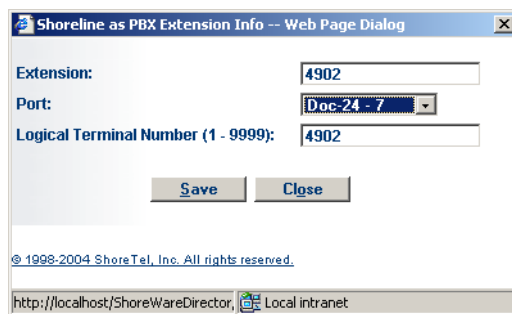


Figure 4-4 External Voice Mail Dialog Box

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

For application servers configured for ShoreTel voice mail, select a translation table from the Translation Table drop-down list. For more information on creating a digit translation table, see Section 2.3 on page 18.

4.5 Fax Servers

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

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

For more information on fax server integration, see the *ShoreTel 8 Planning and Installation Guide*.

Configuring Switches

This chapter provides a general overview of the ShoreGear voice switches as well as information on how to configure them through ShoreWare Director. This chapter provides information on configuring ShoreGear voice switch:

The ShoreGear voice switches provide a highly reliable, highly scalable platform for the distributed call control software. Each ShoreGear voice switch connects to the IP network using a 10/100/1000M Ethernet port.

If more 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 breakpoints with line cards, shelves, cabinets, and systems.

For more information about the features supported outside the U.S. and Canada, refer to the *ShoreTel 8 Planning and Installation Guide*, Appendix A, “International Planning and Installation.”

5.1 Switch Models

ShoreGear models are classified into three switch families, based on chassis type:

- 1-U Half Width Switches
- 1-U Full Width Switches
- IPBX Switches

The following is a brief description of each switch family. Appendix D, starting on page 463, describes each ShoreGear switch, including LED behavior, interface details, capacity, and front panel illustrations.

5.1.1 ShoreGear 1-U Half Width Voice Switches

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

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

ShoreGear 1-U Half Width Voice Switch models include:

- ShoreGear 30 (SG 30)
- ShoreGear 50 (SG 50)
- ShoreGear 90 (SG 90)

- ShoreGear 90BRI (SG 90BRI)
- ShoreGear 220T1 (SG 220T1)
- ShoreGear 220T1A (SG 220T1A)
- ShoreGear T1k (SG T1k)
- ShoreGear 220E1 (SG 220E1)

5.1.2 ShoreGear 1-U Full Width Voice Switches

The ShoreGear 1-U Full Width Switch family includes five models that support analog, IP, SIP, T1, and E1 voice data streams. Full width switch models can be stacked or mounted directly into a standard 19-inch equipment rack. These switches are all 1 RU and have an RJ21X connector for connection to analog phones and trunks. They also feature redundant Ethernet LAN connections for greater availability and reliability.

ShoreGear 1-U Full Width Voice Switch models include:

- ShoreGear 120 (SG 120) – also referred to as ShoreGear 120/24 (SG 120/24)
- ShoreGear 60 (SG 60) – also referred to as ShoreGear 60/12 (SG 60/12)
- ShoreGear 40 (SG 48) – also referred to as ShoreGear 40/8 (SG 40/8)
- ShoreGear T1 (SG T1)
- ShoreGear E1 (SG E1)
- ShoreGear 24A (SG 24A)

5.1.3 ShoreGear IPBX Voice Switches

The ShoreGear IPBX Switch family is the earliest ShoreGear switch design. Although ShoreTel no longer markets IPBX switches, existing switches are supported and can be used in a ShoreTel 6.1 network configuration. IPBX switches are full width and varying heights, depending on the switch model.

ShoreGear IPBX Voice Switch models include:

- ShoreGear 24 (IPBX 24) Voice Switch (2 RU)
- ShoreGear T1 (IPBX T1) Voice Switch (1.5 RU)
- ShoreGear E1 (IPBX E1) Voice Switch (1.5 RU)

5.2 Switch Resources

ShoreGear switches provide telephony, IP phone, and SIP phone resources to ShoreTel 6.1 system users. Each voice switch offers a combination of these resources which can be further customized to support specific individual configurations.

The following sections describe the resources available on ShoreGear voice switches. Individual ShoreGear model descriptions enumerate the features available on the switches.

5.2.1 Analog Circuits

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

- **Extensions:** Extensions are telephony FXS circuits that supply power to connected phones, provide ring signals and dial tone, detect on-hook or off-hook conditions in addition to transmitting and receiving voice signals. Director panels display Extensions as analog ports. Extensions are assigned directly to user extensions.

- **DID Trunks:** DID Trunks handle inbound Loop Reversal trunks that provide DID service from the Central Office. DID trunks are assigned to trunk groups
- **Loop Start Trunks:** Loop start trunks are FXO circuits that handle inbound and outbound from ShorePhones. These trunks accept ring signals, go on-hook and off-hook in addition to transmitting and receiving voice signals. Director panels display Extensions as analog ports. Extensions are assigned to trunk groups.

5.2.2 Digital Circuits

ShoreGear offers T1, E1, and BRI digital circuits that support CAS and ISDN signalling protocols through various 1-U Half Width and 1-U Full Width switches. Circuit channels are configured through Director Switch Edit pages.

5.2.3 IP Phone Ports

ShoreGear switches support varying numbers of ShorePhone IP devices, as specified by the Switch's Edit panel. IP phones resources are allocated as follows:

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

5.2.4 SIP Trunks

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

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

5.2.5 SIP Proxies

ShoreGear switches support varying numbers of SIP trunks, as specified by the Switch's Edit panel. SIP proxies resources are allocated as follows:

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

5.3 Configuration Parameters

Before configuring your switches in ShoreWare Director, you must determine the IP and MAC address assignments for each voice switch. Please refer to the *ShoreTel 8 Planning and Installation Guide* for more information about getting an IP address for each voice switch.

Here are the items that you need before you begin configuring your switches:

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

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

5.3.1 IP Phone, SIP, and Make Me Conference Support

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

- five IP phones
- five SIP trunks
- 100 SIP devices

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

If you do not reserve sufficient ports for IP phones on the switches, some or all IP phones will not be recognized or supported by the ShoreTel 6.1 system. For more information in ShoreTel 6.1 system requirements, see the *ShoreTel 8 Planning and Installation Guide*.

5.3.2 Backup Operator

ShoreGear voice switches feature a backup operator in case the site operator is unreachable due to network outages.

For most switches, the backup operator is on the same port as the Power Fail Transfer port. To use this feature, configure the following ports:

- port 5 on the SG 40
- port 9 on the SG 60 or the SG 120
- port 12 on the SG 30, SG 50, SG 90, SG 220T1A.

5.4 Director Panels

After physically installing a ShoreGear switch into the network, switch parameters are configured through ShoreWare Director panels. This section lists the Director panels that configure and monitor ShoreGear switches.

5.4.1 Switches List Panel

The Switches List panel lists the ShoreGear switches installed in the ShoreTel 6.1 network. To access the Switches List panel, shown in Figure 5-1, select *Administration -> Switches* from the ShoreWare Director menu.

Switches														Help	
Add new switch at site: Headquarters of type: ShoreGear 120/24 Go															
Name	Description	Site	Server	Type	IPAddress	MAC Address	IP Phones In Use	IP Phones Capacity	SIP Trunks In Use	SIP Trunks Capacity	SIP Proxy Capacity	Conference Capacity	Hunt Groups	Music Source	
Klausa	Klausa	Headquarters	Headquarters	SG-90V	10.86.1.127	00-10-49-07-28-0C	0	30	0	0	0	0	0	No	
SG408-JV1	SG408-JV1	Headquarters	Headquarters	408	10.86.3.96	00-10-49-00-29-D6	2	10	0	0	0	0	0	No	
SG408-JV2	SG408-JV2	Headquarters	Headquarters	408	10.86.3.104	00-10-49-00-46-8E	1	5	0	0	0	0	0	No	
SoftSwitch	SoftSwitch	Headquarters	Headquarters	SW	10.1.3.22		0	0	0	0	0	0	0	No	
switch 2		Headquarters	Headquarters	120/24	10.86.107.2	00-10-49-10-72-00	0	0	0	0	0	0	0	No	
switch 2a		Headquarters	Headquarters	SG-24A	10.86.107.1	00-10-49-10-71-00	0	0	0	0	0	0	0	No	
jvds1		Headquarters	jvds1	SW	10.86.0.64		0	0	0	0	0	0	0	No	
Total							3	45	0	0	0	0	0	2	

© 1998-2007 ShoreTel, Inc. All rights reserved.

Figure 5-1 Switches Panel – Director

5.4.1.1 Add new switch at site

The Add new switch at site parameter specifies the site to which a new switch is assigned. To add a new voice switch, select the desired site and type of voice switch, and then click Go. Director responds by displaying the Edit ShoreGear Switch panel for the specified switch.

5.4.1.2 Switch Table contents

Each row corresponds to one switch installed in the ShoreTel 6.1 network. Each column on the table lists a switch parameter as follows:

- **Name:** This is the name of the switch. Clicking a name takes you to the *Switch* edit page, where you can modify the switch configuration.
- **Description:** This describes the switch. It is an optional entry that typically tells where the switch is located or describes how it is used. For example, the switch's description might indicate the wiring closet where the switch is located.
- **Site:** This is the name of the site where the switch is located.
- **Server:** This is the name of server configured to manage the switch.
- **Type:** This is the type of switch.
- **IP Address:** This is the switch's IP address.
- **MAC Address:** This is the switch's Ethernet MAC address.
- **IP Phones In Use:** This is the number of IP phones connected through the switch.
- **IP Phone Capacity:** This is the number of IP phones the switch can support based on the number of ports reserved for IP phones.
- **SIP Trunks In Use:** This is the number of SIP trunks connected through the switch.
- **SIP Trunks Capacity:** This is the number of SIP trunks the switch can support based on the number of ports reserved.
- **Conference Capacity:** The number of ports reserved for Make Me conferences.
- **Hunt Groups:** Shows the number of hunt groups the switch is hosting.
- **Music Source:** This indicates whether there is a music source for music-on-hold.

5.4.2 Switch Edit panel

Switch edit panels configure identification and operating parameters for specified switches installed in the ShoreTel 6.1 network. Director provides a specific panel for each available ShoreGear switch to list only the relevant parameters for an individual switch.

5.4.2.1 New Switch and Edit Switch Panels

The structure of Edit panels generated for adding new switches differs from the structure of Edit panels used to change the settings of existing panels.

Director displays an Edit panel for a new switch, as shown in Figure 5-2, when you add a new switch for the Switches List page.

Port	Port Type	Trunk Group	Description	Jack Number	Location
1	Available				
2	Available				
3	Available				
4	Available				
11	Available				
12	Available				

Figure 5-2 SG 50 Edit Switch panel – Adding a Switch

Edit Switch panels for adding switches typically comprise two sections:

- **The parameter section** is located at the top of the panel and configures identification and signaling settings for the switch. The list of parameters in this section depends on the type of switch being added to the ShoreTel 6.1 network.
- **The port table** is located at the bottom of the panel and configures each port or channel on the switch.

Director displays an Edit page for an existing switch, as shown in Figure 5-4, when you edit an existing switch. To display this switch, click on the name of the desired switch on the Switches List page.

Edit Switch panels for editing existing switches typically comprise three sections.

- **The parameter section** is located at the top of the panel and configures identification and signaling settings for the switch. This section is identical for both types of edit panels.

Switches Help

Edit ShoreGear 50 Switch New Copy Save Delete Reset

Edit this record Refresh this page

Name:

Description:

Site:

IP Address: Find Switches

Ethernet Address:


Server to Manage Switch:

Caller's Emergency Service Identification (CESID): (e.g. +1 (408) 331-3300)

Built-in Capacity: IP Phone + SIP Trunk = Total
 + = 20

Music On Hold Source
 Music On Hold Gain (-49 to 13): dB

Use Analog Extension Ports as DID Trunks



Switch 174

Port	Port Type	Trunk Group	Description	Jack Number	Location
1	Available		P01		
2	Available		P02		
3	Available		P03		
4	Available		P04		
11	Available		P05		
12	Available		P06		

© 1998-2007 ShoreTel, Inc. All rights reserved.

Figure 5-3 SG 50 Edit Switch panel – Editing an Existing Switch

- The **switch graphic** displays the switches front panel between the parameter section and the port table. The Name of the switch is listed below the left side of the graphic. The port popout section provides port configuration information as follows:
 - The color block specifies the port type assignment through the corresponding color blocks. Figure 5-4 provides the code for interpreting these blocks.
 - Hovering the cursor on the LED graphic displays the Port Type setting and the Trunk Group, when appropriate, to which the port is assigned below the left side of the graphic.

	Half Width Switches	Full Width Switches
Trunk	■	■
Extension	■	■
IP phone	■	■
Conference	■	■
SIP proxy	■	■
available	■	■
SIP trunk	■	■

Figure 5-4 Switch Type Color Code

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

- The **port table** is located at the bottom of the panel and configures each port or channel on the switch. This section is identical for both types of edit panels.

5.5 Switch Parameters

Parameters listed on edit pages depend upon the switch that is being edited. The following are lists of parameters that are configured on the various ShoreGear Edit pages.

5.5.1 System Parameters

ShoreGear system parameters configure parameters required by the switch to interact with the ShoreTel 6.1 network. Figure 5-5 displays the portion of a ShoreGear Edit panel that contains these parameters.

The screenshot shows the 'Switches' edit panel for a 'ShoreGear 50 Switch'. At the top, there are buttons for 'New', 'Copy', 'Save', 'Delete', and 'Reset', along with a 'Help' link. Below these is a 'Refresh this page' link. The main form contains the following fields and controls:

- Name:** A text input field containing 'Switch 174'.
- Description:** An empty text input field.
- Site:** A dropdown menu showing 'Headquarters'.
- IP Address:** A text input field containing '10.86.108.3' and a 'Find Switches' button.
- Ethernet Address:** A text input field containing '00-10-49-10-83-00'.
- Server to Manage Switch:** A dropdown menu showing 'Headquarters'.
- Caller's Emergency Service Identification (CESID):** A text input field with a placeholder '(e.g. +1 (408) 331-3300)'.
- Built-in Capacity:** A section with the label 'IP Phone + SIP Trunk = Total' and a form '20 + 0 = 20'.
- Music On Hold Source:** A checkbox that is unchecked.
- Music On Hold Gain (-49 to 13):** A text input field containing '0' and a unit 'dB'.
- Use Analog Extension Ports as DID Trunks:** A checkbox that is checked.

Figure 5-5 System Parameters section of a Switch Edit panel

All ShoreGear switch edit panels list these parameters except as otherwise noted.

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

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

- **Find Switches:** This button invokes the *Find Switch* dialog box. It automatically finds all voice switches of the associated type on the network.

This feature works only when the switch is on the same subnet as the ShoreWare server. The server IP address is given to the voice switch via DHCP/ BOOTP or via a serial port command line entry.

- **Ethernet Address:** This is the switch's Ethernet address. It is the MAC address printed on the switch's rear panel.

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

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

Refer to Appendix B, starting on page 399, for more information.

- **Built-in Capacity:** Built-in capacity allocates switch resources to support IP phones, SIP trunks, and SIP proxies on the ShoreTel 6.1 network. Resource availability varies for each ShoreGear model.

— To allocate IP Phone and SIP Trunk resources, enter the desired number of resources in the data entry boxes.

— To determine the allocated SIP Proxy resources, subtract the number of available resources from the sum of the entered numbers, then multiply the difference by 20.

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

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

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

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

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

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

5.5.2 T1 Signalling Parameters

T1 Signaling parameters configure T1 circuit Layer 3 and Layer 1 parameters. Figure 5-6 displays the portion of a ShoreGear Edit panel that contains these parameters.

The screenshot shows the 'Switches' edit panel for a ShoreGear T1 Switch. The panel includes a header with 'Switches' and 'Edit ShoreGear T1 Switch' text, along with buttons for 'New', 'Copy', 'Save', 'Delete', and 'Reset'. Below the header is a 'Refresh this page' link. The main configuration area is divided into sections:

- General Information:** Name (Switch 202), Description, Site (Headquarters), IP Address (10.86.114.1), Ethernet Address (00-10-49-11-41-00), and Server to Manage Switch (Headquarters).
- Layer 3:** Protocol Type (CAS), Central Office Type (4ESS), and Call by Call Service (<None>). There is also an unchecked checkbox for 'Enable Outbound Calling Name'.
- Layer 1:** Clock Source (Master), Framing Format (D4), Line Code (AMI), and Line Build Out (0.5 dB (0 - 110 feet)).

Figure 5-6 T1 Parameters section of a Switch Edit panel

Panels that configure the following parameters include the SG T1, SG 220 T1, and the SG 220T1A.

5.5.2.1 Layer 3

Layer 3 parameters configure the following network layer settings.

- **Protocol Type:** This determines the type of signaling the ShoreGear-T1 will facilitate:
 - *CAS* uses in-band signaling where a portion of the bearer channel is used for A/B bit signaling, emulating on-hook and off-hook conditions.
 - *ISDN User* and *ISDN Network* is used in conjunction with ISDN signaling configurations. It provides a single D-channel for controlling 23 bearer channels. You can designate the T1 trunk as being *User* or *Network* side.
 - The *QSIG protocol* handles the signaling between digital PBXs for ISDN, handling the basic signaling functions of call setup, handshaking, and call teardown, and thus allowing PBXs from different third-party vendors to communicate and interoperate.

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

- Select **QSIG Master** or **QSIG Slave** from the drop-down menu, depending on which type of signaling the ShoreGear-T1 will facilitate.
- **Central Office Type:** This provides support for the following types of Central Offices (COs):
 - 4ESS
 - 5ESS
 - DMS-100
 - National ISDN-2 (NI-2)

Select an appropriate CO type from the drop-down list. The default is **4ESS**.

- **Call by Call Service:** Call by Call Service is a 4ESS feature that allows a user to access different services, such as an 800 line or WATS line, on a per call basis. This parameter is available only when Central Office Type is set to 4ESS.
Parameter options (SDN or MEGACOM) are AT&T outbound service types.
- **Enable Outbound Calling Name:** This sends the caller name with the caller ID for outbound calls. The default is disabled.

5.5.2.2 Layer 1

Layer 1 parameters configure the following physical layer settings.

- **Clock Source:** This is the ShoreGear-T1 switch's clock source. Depending on the type of T1 service provided by your telephone company's CO, select either **Slave** or **Master** from the drop-down list. Typically the ShoreGear-T1 is slave to the central office. The system default is **Slave**.
- **Framing Format:** This is the ShoreGear-T1 switch's framing format. Depending on the type of T1 service provided by your telephone company's CO, select either **ESF** or **D4** from the drop-down menu. The system default is **ESF**.
- **Line Code:** This is the ShoreGear-T1 switch's line code. Depending on the type of T1 service provided by your telephone company's CO, select either **B8ZS** or **AMI**. The system default is **B8ZS** from the drop-down menu.
- **Line Build Out:** This drop-down list provides a list of T1 trunk line distances, specified in decibels (dB) and in feet. Select the appropriate line code from the drop-down list.

5.5.3 E1 Signalling Parameters

E1 Signalling parameters configure E1 circuit Layer 3 and Layer 1 parameters. Figure 5-7 displays the portion of a ShoreGear Edit panel that contains these parameters.

Panels that configure the following parameters include the SG E1, and the SG 220E1.

5.5.3.1 Layer 3

Layer 3 parameters configure the following network layer settings.

- **Protocol Type:** This parameters specifies the signalling protocol.
 - **ISDN User** and **ISDN Network** are ISDN signalling protocols.
 - **QSIG** is an ISDN based signalling protocol used for signalling between PBXs in a private network.

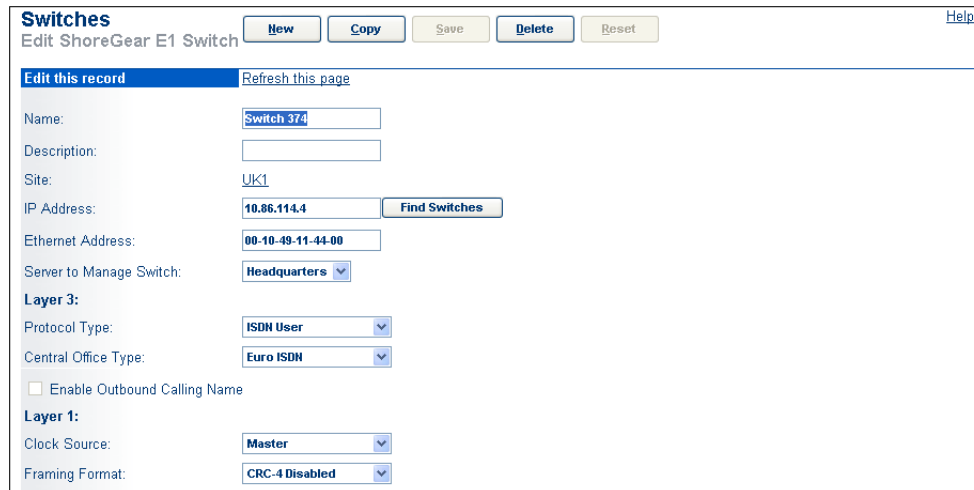


Figure 5-7 E1 Parameters section of a Switch Edit panel

- **Central Office Type:** The ShoreGear-E1 supports a single signaling type per country, which is typically *Euro-ISDN(TBR4)*. This parameter is active only if Protocol Type is set to *ISDN User* or *ISDN Network*.
- **Enable Outbound Calling Name:** When this parameter is selected, the caller name is sent with the caller ID for outbound calls. The parameter is disabled by default.

5.5.3.2 Layer 1

Layer 1 parameters configure the following physical layer settings.

- **Clock Source:** This parameter specifies the clock source. This parameter is typically set for compatibility with the service provided by the telephone company's CO. The default setting is *Slave*.
- **Framing Format:** The ShoreGear-E1 supports *CRC-4*.

5.5.4 BRI Signalling Parameters

BRI Signaling parameters configure BRI circuit Layer 1, Layer 2, and Layer 3 parameters. Figure 5-8 displays the portion of a ShoreGear Edit panel that contains these parameters.

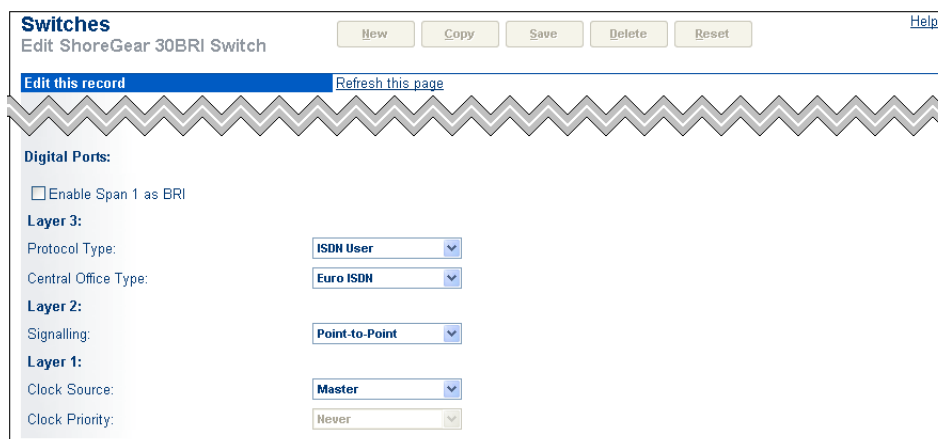


Figure 5-8 BRI Parameters section of a Switch Edit panel

Switch edit panels provide a signalling parameter set for each BRI span on the ShoreGear Switch. The ShoreGear 90BRI supports four BRI spans; the ShoreGear 30 BRI supports one BRI span.

5.5.4.1 Enable Span as BRI

Select this option to Activate the corresponding Digital Channels as a BRI span. When this parameter is not selected, the Channel resources are available for reallocation to support IP phones, SIP trunks, or SIP proxies.

5.5.4.2 Layer 3

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

5.5.4.3 Layer 2

- **Signaling:** Select either **Point-to-Point** or **Point-to-Multipoint**.

5.5.4.4 Layer 1

- **Clock Source:** This parameter specifies the clock source. This parameter is typically set for compatibility with the service provided by the telephone company's CO. The default setting is *Slave*.
- **Clock Priority:** This priority configures the clock recovery priority for the BRI voice channels.
 - When the Clock Source for the ShoreGear 90BRI is set to **Master**, the Clock Priority is set to **Never**.
 - When the Clock Source is set to **Slave**, this parameter configure the priority level: **High**, **Medium**, **Low**, or **Lower**.

This parameter has no significance for the SG 30BRI, regardless of the Clock Source setting.

5.5.5 Analog Port Tables

Analog Port Tables are located below the System Parameters section on Switch Edit panels for switches that provide analog circuit resources. Figure 5-9 displays an analog port table for an SG 90 switch.

- **Port Type**—This parameter configures the port resources.
 - *Available* configures the port resources to support either an extension port or DID trunk. Port capabilities depend on the ShoreGear switch model.
 - *Conference* configures the port resources for Make Me conferencing.

Reserve as many ports as you need to support the maximum number of conferences you will permit to occur simultaneously. For example, if you will

Port	Port Type	Trunk Group	Description	Jack Number	Location
1 Edit	Trunk	Analog Loop Start	P01		
2	Conference		P02		
3 Edit	Trunk	Analog Loop Start	P03		
4	Conference		P04		
5	Conference		P05		
6	Conference		P06		
7	5 IP Phones		P07		
8 Edit	Trunk	Analog Loop Start	P08		
9	5 SIP Trunks		P09		
10	Available		P10		
11	Available		P11		
12	Available		P12		

Figure 5-9 Analog Port Table – SG 90 Edit panel

allow two three-way calls at the same time, reserve 6 ports. The minimum number of ports that can be reserved for Make Me conferencing is four.

- *Trunk* configures the port as an analog trunk assigned to the Trunk Group specified by the Trunk Group parameter.
- *5 IP Phones* configures the resource to support 5 IP phones.
- *5 SIP Trunks* configures the resource to support 5 SIP trunks.
- *100 SIP Proxy* configures the resource to support 100 SIP proxies.
- **Trunk Group**—This parameter specifies the Trunk Group to which the port is assigned. This parameter is available only when Port Type is set to Trunk.

Port tables do not include the Trunk Group column for switches that do provide Loop Start Trunks.
- **Description**—This parameter is an optional comment field that lists a descriptive name for the switch port.
- **Jack Number**—This parameter is an optional comment field for entering the patch-panel jack number to which the port is connected.
- **Location**—This parameter is an optional comment field for entering location information about the port.

5.5.6 Digital Channel Tables

Digital Port Tables are located below the System Parameters section on Switch Edit panels for switches that provide digital analog circuit resources. Figure 5-10 displays an analog port table for an SG 220T1 switch.

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

Switches
Edit ShoreGear 220T1 Switch

New Copy Save Delete Reset Help

Channel	Port Type	Trunk Group	Description	Jack Number	Tx Gain (dB)	Rx Gain (dB)
1 Edit	Trunk	Q1 PRI Tie	Channel		0	0 Fill Down
2 Edit	Trunk	Q1 PRI Tie	Channel (2)		0	0
3 Edit	Trunk	Q1 PRI Tie	Channel (3)		0	0
4 Edit	Trunk	Q1 PRI Tie	Channel (4)		0	0
5 Edit	Trunk	Q1 PRI Tie	Channel (5)		0	0
6 Edit	Trunk	Q1 PRI Tie	Channel (6)		0	0
7 Edit	Trunk	Q1 PRI Tie	Channel (7)		0	0
8 Edit	Trunk	Q1 PRI Tie	Channel (8)		0	0
9 Edit	Trunk	Q1 PRI Tie	Channel (9)		0	0
10 Edit	Trunk	Q1 PRI Tie	Channel (10)		0	0
11 Edit	Trunk	Q1 PRI Tie	Channel (11)		0	0
12 Edit	Trunk	Q1 PRI Tie	Channel (12)		0	0
13 Edit	Trunk	Q1 PRI Tie	Channel (13)		0	0
14 Edit	Trunk	Q1 PRI Tie	Channel (14)		0	0
15 Edit	Trunk	Q1 PRI Tie	Channel (15)		0	0
16 Edit	Trunk	Q1 PRI Tie	Channel (16)		0	0
17 Edit	Trunk	Q1 PRI Tie	Channel (17)		0	0
18 Edit	Trunk	Q1 PRI Tie	Channel (18)		0	0
19 Edit	Trunk	Q1 PRI Tie	Channel (19)		0	0
20 Edit	Trunk	Q1 PRI Tie	Channel (20)		0	0
21 Edit	Trunk	Q1 PRI Tie	Channel (21)		0	0
22 Edit	Trunk	Q1 PRI Tie	Channel (22)		0	0
23 Edit	Trunk	Q1 PRI Tie	Channel (23)		0	0
24 Edit	Unavailable	Unavailable			0	0

© 1998-2007 ShoreTel, Inc. All rights reserved.

Figure 5-10 Digital Port Table – SG 220T1 Edit panel

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

5.6 Adding a ShoreGear Switch

The following procedure adds a ShoreGear switch to the ShoreTel 6.1 system

Step 1 Access the Switches panel in Director by selecting *Administration -> Switches* from the Director menu.

Step 2 Select the desired site and switch type in the **Add new switch at site** parameters located at the top of the panel, then click *Go*.

Director displays the Switch Edit panel for the desired switch.

Step 3 Enter the system parameter settings in the data entry fields at the top of the panel. Refer to Section 5.5.1 on page 62 for a description of the system parameters.

Step 4 If the switch will be used as a music-on-hold source, enable it by clicking the *Music Source* check box.

Step 5 If your DHCP/ BOOTP server is running, click **Find Switches** and select an IP address from the **Find Switch** dialog box. This also adds the switch's MAC address in the **Ethernet Address** field.

The Find Switches feature will only find switches on the subnet of the server. You must manually type in the IP and Ethernet address for remote switches.

If your DHCP/ BOOTP server is not running, enter the switch's IP address manually in the *IP Address* field and enter the switch's MAC address in the *Ethernet Address* field.

Step 6 For ShoreGear switches that support digital circuits, enter the required settings in the Layer 1, Layer 2, and Layer 3 data entry fields.

Refer to Section 5.5.2 on page 64 for information on T1 data entry fields.

Refer to Section 5.5.3 on page 65 for information on E1 data entry fields.

Refer to Section 5.5.4 on page 66 for information on BRI data entry fields.

Step 7 For ShoreGear switches that support analog ports, select the desired port type and trunk group options in the Analog Port Table, then complete the remaining data entry fields.

Refer to Section 5.5.5 on page 67 for information on Analog Port Table parameters.

Step 8 For ShoreGear switches that support digital channels, select the desired port type and trunk group options in the Digital Channel Table, then complete the remaining data entry fields.

Refer to 5.5.6 on page 68 for information on Analog Port Table parameters.

Step 9 Click *Save* to save the configuration. The new switch is added to the table in the *Switches* page. If you modified an existing switch configuration, the table will be updated.

When adding a ShoreGear voice switch to a network server you may need to manually power-cycle the voice switch to cause a software upgrade.

5.7 SoftSwitch

SoftSwitch is used to host virtual users who are not assigned a physical telephone port on any ShoreGear voice switch. The SoftSwitch is used to host all voice mail, auto-attendant, and workgroup extensions as well as route points. A SoftSwitch is automatically created for every server added to the ShoreTel 6.1 system. The server will be listed as a SoftSwitch on the Switches list page.

To access the *SoftSwitch* edit panel, as shown in Figure 5-11, click the name of the SoftSwitch from the *Switches* list panel.

The screenshot shows the 'Switches' edit panel for a SoftSwitch. At the top, there are 'Save' and 'Reset' buttons and a 'Help' link. Below that is a 'Refresh this page' link. The main form contains the following fields:

- Name: SoftSwitch 1
- Description: SoftSwitch 1
- Site: Headquarters
- IP Address: 10.1.3.22

There is a link 'Go to Server page to change IP Address' next to the IP address field. At the bottom, there is a copyright notice: © 1998-2007 ShoreTel, Inc. All rights reserved.

Figure 5-11 SoftSwitch Edit Panel

The parameters that appear on the *SoftSwitch* edit page are as follows:

- **Name:** This is the name of the SoftSwitch. The default name is SoftSwitch, but you can change the name to suit your system.
- **Description:** This is a descriptive name of the SoftSwitch.
- **Site:** This is the location of the SoftSwitch. This is a read-only parameter, and cannot be changed. The SoftSwitch is always located at the main site.
The name of the main site is defaulted to “Headquarters,” but can be changed via the Site edit page.
- **IP Address:** This read-only field is the IP address of the switch supporting the SoftSwitch. Change the IP address from the *Server* page.

5.8 Replacing a ShoreGear Voice Switch

The following procedure is recommended for replacing a ShoreGear voice switch:

- Step 1** Make sure you are ready to deliver an IP address to the new voice switch.
If you are using DHCP, the voice switch will request a new IP address at power up.
If you are using the BOOTP server on the ShoreWare server, create a new entry on the BOOTP server page.
- Step 2** Physically install the new voice switch. Power-on the new voice switch. A new IP address will be requested by the switch.
- Step 3** In ShoreWare Director, open the *Switch* edit page for the existing voice switch that is being replaced. Click *Find Switches* to discover the new voice switch. Select the voice switch and save.

If the switch is on a different subnet of the ShoreWare server, Find Switches will not work and you must manually type in the IP and MAC addresses.

- Step 4** Unplug the port connections (telephones, trunks) from the existing voice switch and plug them into the new voice switch.
- Step 5** In ShoreWare Director, go to the *Quick Look* page to confirm that the voice switch comes back on line. This could take one or two minutes.

Voicemail Model Switches

6.1 Introduction

6.1.1 Definition

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

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

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

Three ShoreGear switches operate as a ShoreGear voice switch and as a voice mail server:

- ShoreGear 90V
- ShoreGear 90BRIV
- ShoreGear 50V

6.1.2 Background

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

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

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

6.2 Functional Description

6.2.1 Physical Description

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

6.2.1.1 Capacity

The following are V Model switch capacities:

System

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

ShoreGear 90V

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

ShoreGear 90BRIV

- Analog Telephony: 4 extension ports
- Digital Telephony: 4 BRI ports – 8 channels
 - each port supports one BRI span that comprises two channels
- IP and SIP resources
 - 90 IP phones, 90 SIP trunks, or 1800 SIP proxies maximum – requires reallocation of telephony resources.
 - 30 IP phones, 30 SIP trunks, or 600 SIP proxies independent of telephony support
- 90 voice mailboxes

ShoreGear 50V

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

6.2.1.2 Differences from other Switches

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

- V Model switches have a slot on the left side of the chassis for accessing the CF card.
- V Model switches provide Voicemail and auto attendant services normally provided by the Main Server or a Distributed Server.
- V Model switch and server functions run on Linux.
The OS on other ShoreGear switches is VxWorks. Other ShoreWare Servers run under Microsoft Windows.
- V Model switches do not support Simplified Message Desk Interface (SMDI).

6.2.2 Voice Switch Functions

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

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

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

6.2.3 Server Functions

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

6.2.3.1 Voicemail

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

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

Voicemail box capacity varies by switch model:

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

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

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

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

6.2.3.2 Auto Attendant Menus

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

6.2.3.3 Recorded Name Storage

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

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

6.2.3.4 Voicemail Prompts

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

6.2.4 Connectivity Requirements

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

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

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

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

6.3 Utilities

6.3.1 Accessing Voicemail Model Switch Utilities

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

For security purposes, the Switch accepts requests from only CLIs s running on the local host, the controlling Distributed server, or the Main ShoreTel server.

6.3.1.1 Accessing Utilites from the Serial Port

Switch utilities are accessible through the maintenance port located on the faceplate. The available utility depends on the state of the switch at the time of access.

- During normal switch operation, the Maintenance port provides access to a specified Linux shell. By default, STCLI is the specified shell.

To access ShoreGear utilities through the Maintenance:

Step 1 Connect a serial cable between a personal computer and the ShoreGear switch.

Step 2 Use a terminal emulation program, such as Tera Term Pro freeware, to open a connection to the switch.

Step 3 Apply these values to the terminal settings:

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

Refer to utility descriptions for access information through the Maintenance port.

6.3.1.2 Accessing Utilities from SSH

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

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

Logging into the Admin account immediately opens the STCLI interface.

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

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

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

To access a Linux BASH Shell account:

Step 1 Open a SSH client access panel.

Figure 6-1 displays the PuTTY Configuration panel.

Step 2 Open the command line by entering the IP address of the switch to which the command line will access in the appropriate data entry field, then pressing the button that activates the command shell.

In Figure 6-1, the IP address is placed in the Host Name (or IP address) field, after which pressing the Open button opens the command line interface.

Step 3 Enter the username of the desired account on the command line, then press Enter key.

In Figure 6-1, admin is the specified account.

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

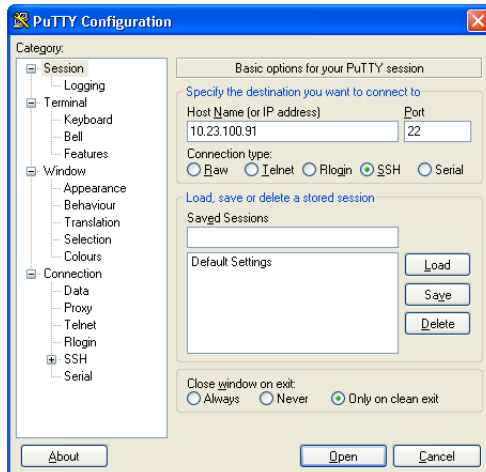


Figure 6-1 PuTTY Configuration panel

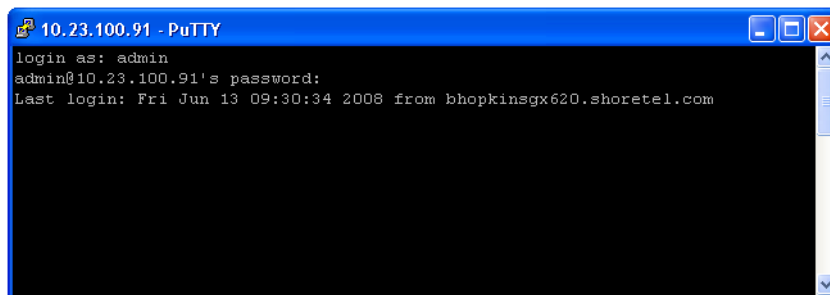


Figure 6-2 PuTTY Command Prompt window

6.3.2 Switch Utilities

A set of command line interface (CLI) commands supports various operations. The commands to backup and restore voicemail, stop or start services and groups, format and CF cards, among other operations.

6.3.2.1 STCLI

STCLI, also known as the ShoreTel Shell, displays and modifies system configuration parameters. You can implement static or dynamic IP addressing for the switch from STCLI. You can also reboot the switch from STCLI. Figure 6-3 displays the STCLI prompt. The main STCLI menu appears below the ShoreTel logo.

To open STCLI:

- Access the Maintenance port, as described in Section 6.3.1.1. If window displays the Linux prompt, enter STCLI on the command line.
- Access the SSH and log in as the administrator, as described in Section 6.3.1.2.

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


```

login as: admin
admin@10.23.100.117's password
Last login: Mon Mar 3 17:52:45 2008 from dsanten-1808.shoretel.com
Flags=0x40
[admin@Kilauea ~]$ stcli

      .sssS Ssss,
      sSSSSSSSSSSSSs,
      SSSS~ ~SSS,
      SSSS _sSSSSSSSS SS
      sSSS SSSSSSSSSSs
      SSSSs SSSS~SSSSSSs
      'SSSS ~ _s, ~SSSS
      SSSSSSSs_sSSSS }SSSS
      ~SSSSSSSSSS' }SSS'
      Ss ~SSSSSSS~ SSSS
      'SSs, _sSSSS
      ~SSSSSSsSSSSSS^
      ~SSSSSS~^

Commands for ShoreGear:
(0) -- Exit
(1) -- Show version
(2) -- Show system configuration
(3) -- Change system configuration
(4) -- Reboot
? -- Help

ShoreTel>

```

Figure 6-3 STCLI Login and Main Menu

6.4 V Model Switch Implementation

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

- Adding a new V model switch to a ShoreGear server, described in Section 6.4.1
- Configuring voice mail, described in Section 6.4.4
- Specifying Linux root and administrator passwords, described in Section 6.4.5
- Specifying maximum size and age of log files stored on the CF card, described in Section 6.6.2
- Configuring backup of voice mail and enabling its automatic activation, described in Section 6.4.6.

Monitoring memory usage on the CF card, described in Section 6.6.3.

6.4.1 Installing a Switch

After physically connecting a V Model Switch into the system, the switch is added to the system through Director. Switch previously configured through Director appear in status and configuration panels for both Switches and Application Servers.

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

Step 1 Select *Administration -> Switches* from the Director Main Menu

Director displays the Switches panel, as shown in Figure 6-4.

Step 2 Assign the switch to a site from the **Add new switch at site** drop down menu in the top left corner of the window.

Step 3 Select the switch model in the **of type** drop down menu, located right of the **Add new switch at site** menu.

In Figure 6-4, a ShoreGear 90V switch is being added.

Switches								
Add new switch at site: <input type="text" value="HQ"/> of type: <input type="text" value="ShoreGear 90V"/> <input type="button" value="Go"/>								
Name	Description	Site	Server	Type	IPAddress	MAC Address	IP Phones In Use	IP Ph Capa
E911		HQ	Headquarters	T1	10.86.26.106	00-10-49-03-B8-78	0	
Floater		HQ	Headquarters	T1	10.86.0.207	00-10-49-05-92-BF	0	
SG90VBR1		UK site	Headquarters	90BRIV	10.23.87.105	00-10-49-08-14-D8	2	
Swarna T1		HQ	Headquarters	T1	10.23.79.207	00-10-49-00-42-92	0	
HQVMB-50V	50V	HQ	Headquarters	SG-50V	10.23.86.106	00-10-49-08-1D-F0	2	
RemVMB90V	90V	Remote	RemoteAPPserv	SG-90V	10.86.86.119	00-10-49-07-28-02	0	
HQ220T1A	HQ220T1A	HQ	Headquarters	SG-220T1A	10.23.86.50	00-10-49-07-28-29	0	
HQFuji120	HGFuji120	HQ	Headquarters	120/24	10.23.86.30	00-10-49-04-98-A0	1	
HQFuji60	HGFuji60	HQ	Headquarters	60/12	10.23.86.40	00-10-49-00-50-76	2	
Rem220T1	Rem220T1	Remote	RemoteAPPserv	SG-220T1	10.23.79.229	00-10-49-07-27-AB	1	
REM50	REM50	Remote	RemoteAPPserv	SG-50	10.86.86.45	00-10-49-07-27-DD	0	
REMFuji120	REMFuji120	Remote	RemoteAPPserv	120/24	10.23.86.35	00-10-49-00-3F-C0	1	
RemoteAPPserv	RemoteAPPserv	Remote	RemoteAPPserv	SW	10.23.86.20		0	
RemSG30	RemSG30	Remote	RemoteAPPserv	SG-30	10.23.87.102	00-10-49-09-60-13	0	
SoftSwitch	SoftSwitch	HQ	Headquarters	SW	10.23.86.15		0	
Remote90V	VMB	Remote	RemoteAPPserv	SG-90V	10.23.86.75	00-10-49-07-28-04	3	
Total							12	

© 1998-2008 ShoreTel, Inc. All rights reserved.

Figure 6-4 Adding a ShoreGear 90V V Model Switch

Step 4 Click the **Go** located right of the type drop down menu. to open the configuration window for the new switch. Figure 6-5 and Figure 6-6 display the fields required for configuring a V model type of switch.

Step 5 Complete each field according to the plan for this switch and network.

Refer to Section 5.4 on page 58 for instructions on configuring a ShoreGear Voice Switch.

The switch configuration window for V model switches contains voice mail and backup options in addition to voice switch options available for other switches. Figure 6-5 shows the upper half of the panel that configures switch parameters.-related part of the screen.

Figure 6-6 shows the lower half of the panel that configures , voicemail and automatic backup. Refer to Section 6.4.4 on page 82 for instructions on configuring voice mail. Refer to Section 6.4.6 on page 84 for instructions on configuring system back-up.

6.4.2 Replacing a Switch

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

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

Step 1 Remove the original switch from the ShoreTel network.

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

Step 3 Remove the CF card from the memory slot.

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

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

Switches Help

Edit ShoreGear 50V Switch New Copy Save Delete Reset

Edit this record Refresh this page

Name:

Description:

Site:

IP Address: Find Switches

Ethernet Address:


Server to Manage Switch:

Caller's Emergency Service Identification (CESID): (e.g. +1 (408) 331-3300)

Built-in Capacity: IP Phone + SIP Trunk = Total
 + = 19 of 20 (20 SIP proxy ports)

Music On Hold Source
 Music On Hold Gain (-49 to 13): dB

Use Analog Extension Ports as DID Trunks



Port	Port Type	Trunk Group	Description	Jack Number	Location
1	Available		P01		
2	5 IP Phones		P02		
3	5 IP Phones		P03		
4	5 IP Phones		99621013		
11	Extension		HQ50VAP D		
12	Available		p06		

Voice Mail and Auto-Attendant:

© 1998-2008 ShoreTel, Inc. All rights reserved.

Figure 6-5 Configuring a New V Model Switch

Step 6 Connect the replacement switch into the network.

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

Step 8 Open the Edit ShoreGear Switch panel by clicking the name of the replaced switch.

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

6.4.3 Upgrading a Switch

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

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

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

6.4.4 Configuring Voice Mail

Voicemail is configured through one of the following Director panels:

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

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

- The V model includes the enable and configuration of automatic daily backup.
- Simplified Message Desk Interface (SMDI) is disabled.

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

The screenshot shows the 'Application Servers' configuration window for 'HQVMB-50V'. The window includes a 'Save' button, a 'Reset' button, and a 'Help' link. The configuration is organized into several sections:

- Edit this record:** Refresh this page
- Name:** HQVMB-50V
- Host IP Address:** 10.23.86.106
- Site:** HQ
- SoftSwitch Name:** HQVMB-50V
- Maximum Trunks for Voice Mail Notification (1 - 200):** 4
- Voice Mail and Auto Attendant:**
 - Voice Mail Extension:** 65333
 - Voice Mail Login Extension:** 65339
 - Auto Attendant Extension:** 65335
 - Assigned User Group:** Voice Mail Notification
 - Default Auto-Attendant Menu:** Default (Menu) - 65700
- Backup:**
 - Enable Daily Backup
 - IP Address:** 10.23.86.15
 - FTP Port:** 21
 - Directory:** (empty field)
 - User ID:** anonymous
 - Password:** (two masked password fields)
- Simplified Message Desk Interface:**
 - Mode:** <None>

© 1998-2008 ShoreTel, Inc. All rights reserved.

Figure 6-6 Configuring Voice Mail through Application Servers Window

- **Edit Switches**, accessed by selecting *Administration -> Switches*, then selecting the desired V Model switch.

Figure 6-7 displays the Voicemail parameters on the Switches window for the V model switch named HQVMB-50.

Director paths that open voice mail on the Main and Distributed Servers also access panels that configure V model switches. For details on voice mail configuration, see Chapter 4.

6.4.5 Specifying Root and Administrator Passwords

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

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

Switches
Edit ShoreGear 50V Switch

[New](#) [Copy](#) [Save](#) [Delete](#) [Reset](#) [Help](#)

Edit this record [Refresh this page](#)

Name:
 Description:
 Site:
 IP Address: [Find Switches](#)

Voice Mail and Auto-Attendant:

Maximum Trunks for Voice Mail Notification (1 - 200):
 Voice Mail Extension:
 Voice Mail Login Extension:
 Auto-Attendant Extension:
 Assigned User Group:
 Default Auto-Attendant Menu:

Backup:

Enable Daily Backup
 IP Address:
 FTP Port:
 Directory:
 User ID:
 Password:

© 1999-2008 ShoreTel, Inc. All rights reserved.

Figure 6-7 Configuring Voice Mail through the Switches Window

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

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

Director provides utilities to change the CLI passwords. Modifying the passwords will prevent unauthorized access to internal switch code.

After introducing the two CLIs, this section describes how to specify the passwords:

To change the Voicemail Switch passwords:

Step 1 Select *Administration* → *System Parameters* → *Other* from Director Main Menu.

The “admin” and “root” password fields appear under the ShoreGear Passwords heading in Figure 6-8.

Step 2 Enter new passwords in the admin and root data entry fields.

The character range for both passwords is 4 to 26 ASCII characters.

Step 3 Repeat the passwords in the adjacent data entry fields.

System Parameters
Edit Other Parameters

Save Reset Help

Edit this record Refresh this page

Passwords:
 Max Voice Mail Errors (2-50):
 Min Voice Mail Password Length (4-26):
 Min Client Password Length (4-26):
 Max Client Password Length (4-26):

ShoreGear Voice Switches Passwords:
 "admin" Password:
 "root" Password:

Log File Storage:
 Max Days (1-30):
 Max Size (10-30000): MB

ShoreGear Voice Switches Log File Storage:
 Max Days (1-30):
 Max Size (10-16000): MB

General:
 Prevent Users from Initiating Client Upgrades

© 1998-2008 ShoreTel, Inc. All rights reserved.

Figure 6-8 Specifying Voicemail Switch Parameters

6.4.6 Voicemail Backup

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

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

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

Automatic Backup can be configured on the following Director windows:

- **Edit Application Servers**, accessed by selecting *Administration -> Application Servers*, then selecting the desired V Model Switch. Refer to Figure 6-6.
- **Edit Switches**, accessed by selecting *Administration -> Switches*, then selecting the desired V Model switch. Refer to Figure 6-7.

To configure Automatic Backup in Director:

Step 1 Open the Edit Application Servers or Edit Switches panel for the desired V Model Switch.

Step 2 Place a mark in the **Enable Daily Backup** checkbox.

Step 3 Enter the IP address for the FTP server that will receive the backup files in the **IP Address** data entry field.

Step 4 Enter the port number of the recipient FTP server in the **FTP Port** data entry field.

Step 5 Type the user name for the FTP server account that will receive the backup files..

Step 6 Type password required to authenticate access to the FTP server account.

Step 7 Specify the directory on the FTP server that will receive the backup files..

The screenshot shows the 'Application Servers' configuration page in Director. The 'Backup' section is highlighted, showing the following configuration:

- Enable Daily Backup
- IP Address: 10.23.86.15
- FTP Port: 21
- Directory: (empty field)
- User ID: anonymous
- Password: (masked with asterisks)
- Simplified Message Desk Interface Mode: <None>

Figure 6-9 Voice Mail and Backup Configuration

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

6.4.7 Configuring a Target Server for Backup

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

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

In the following summarizes configuration data for the FTP Server entered into the Backup parameter data entry fields in Director:

- Enable Daily Backup (check mark in box)
- FTP Server: 10.1.1.42
- FTP Port: 5555
- FTP User: backup-10.1.3.93
- FTP Password: backup
- FTP Directory: Backup-10.1.3.93

The following summarizes Switch information entered in FTP Server configuration fields:

- The IP address of the V model switch is 10.1.3.93.
- The IP address of the FTP server is 10.1.1.42, and the port number is 5555.
- The local path to specify in the FTP site directory is C:/ShoreTelBackup.
- FTP user isolation is specified.
- The V model switch user-name sent to the FTP server is “Backup-10.1.3.93;” the user-password is “backup.”

When this configuration data is provided to the FTP server, the following path is created:

C:\ShoreTelBackup\LocalUser\Backup-10.1.3.93

On the server, perform the following:

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

In Figure 6-10, an FTP site named “ShoreTelBackup” is added.

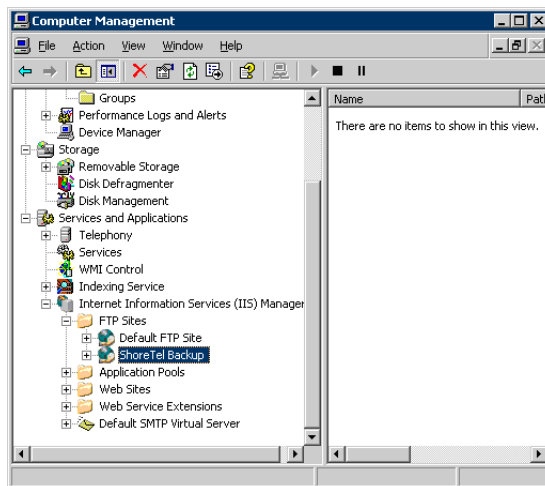


Figure 6-10 The New Target Server is Identified

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

In Figure 6-11, the IP address is 10.1.1.42. and port is 5555.

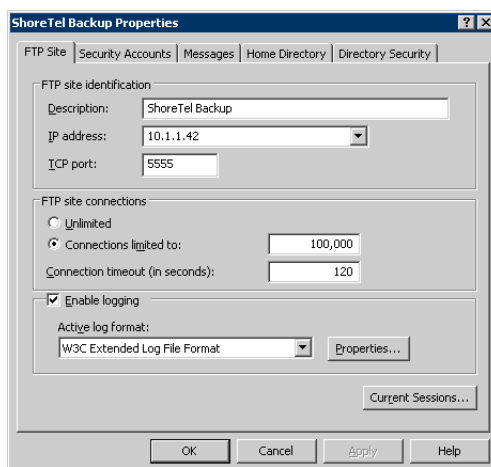


Figure 6-11 Specify IP Address and Port Number for “ShoreTelBackup

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

In Figure 6-12, the local path is C:\ShoreTelBackup.

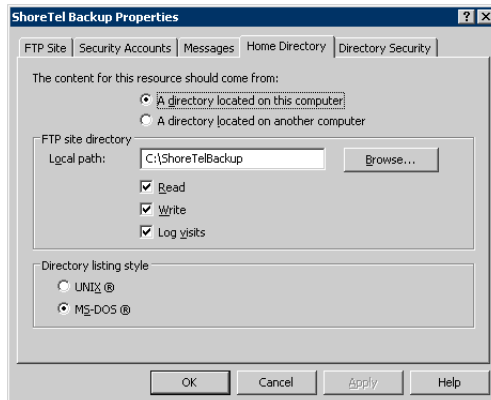


Figure 6-12 Specifying the Local Path for the Backed-up Data

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

Figure 6-13 displays a properly configured Site Creation Wizard.

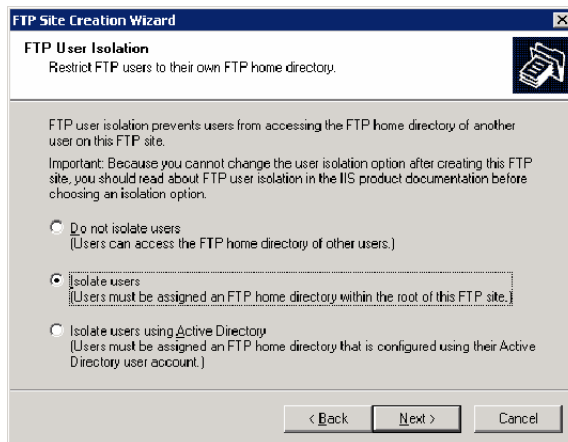


Figure 6-13 Isolating the User (Directory)

Step 5 Enter a name and description of the FTP server in the General panel, as shown in Figure 6-14.

Step 6 Enable the following options:

- User cannot change password
- Password never expires.

Step 7 Verify the resultant path of the configured server, as shown in Figure 6-15.

6.5 Booting and Restarting

6.5.1 Description

Rebooting and restarting V model switches have different scopes.

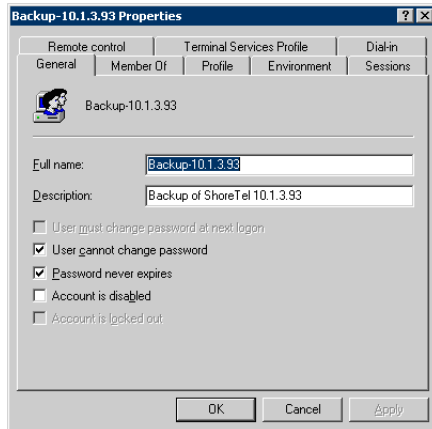


Figure 6-14 Specifying the and Password Details

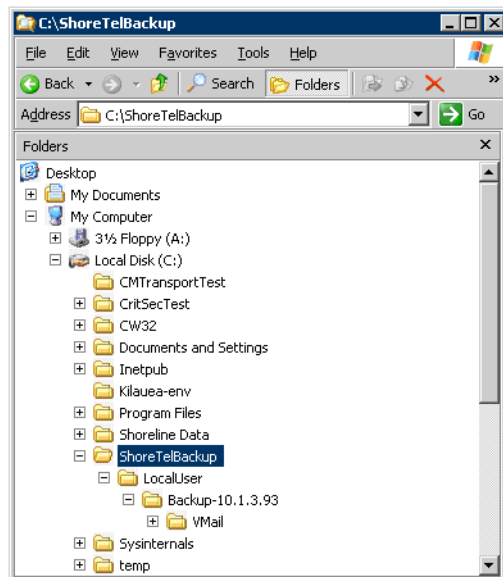


Figure 6-15 After Configuring the FTP Server is Finished

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

A reboot takes much longer than a restart.

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

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

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

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

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

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

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

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

6.5.2 Specifying a Static IP Address

V Model switches normally use DHCP to dynamically set the device IP address and specify the addresses of the servers to which it communicates. Switches are set into fixed address mode through CLI instructions.

The following procedure places the V Model switch into fixed address mode:

Step 1 Access the STCLI command line interface, as described in Section 6.3.2.1 on page 78.

Step 2 Enter “3” on the command line to select **Change System Configuration**.

The CLI window displays the Change System Configuration options.

Step 3 Enter “6” on the command line to select **Enable/Disable DHCP/BOOTP**

The CLI window displays the DHCP/BOOT options.

Step 4 Enter “0” on the command line to select **Manual Configuration**.

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

Step 6 After completing changes to the configuration, type **Exit** to close the CLI.

Step 7 Reboot the switch.

6.5.3 Specifying a Time Source

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

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

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

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

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

6.5.4 Reboot Methods

6.5.4.1 Flash Boot

The standard method for booting a ShoreGear voice switch is to boot from the switch's flash memory. When a ShoreGear switch is first powered on, it reads the boot parameters stored on the boot ROM, which instructs the switch to load software from flash memory. When the software starts, it loads its configuration, which is also stored in flash memory.

6.5.4.2 Default Button

The Default Button is the small “paperclip” button on the switches. Pressing this button will invoke scripts that will overwrite the contents of the NAND flash with the contents from the FTP Server. The contents of the Compact Flash are untouched.

Pressing this button and holding for 10 seconds will also reformat the Compact Flash.

6.5.4.3 FTP Boot

Booting from FTP is available for instances when you cannot boot the switch from flash memory. When you boot a switch from FTP, the operating system and software load from the FTP site identified in the boot parameters. The loaded configuration is a configuration received from the TMS server.

Voicemail services on the switch are disabled after booting from FTP and are restarted only by booting from Flash. After an FTP boot, the switch can perform telephony functions as those available through other ShoreGear switches.

FTP boot on a V model switch is used for troubleshooting. FTP boot can also support certain maintenance tasks and the backup and restore facilities. However, a V model switch that has been booted by way of FTP cannot run voicemail: it can operate only as a voice switch (controlling phones, trunks, and call routing).

6.6 Monitoring a Voicemail Model Switch

6.6.1 Monitoring Switch Functions

Quick Look, which is available from ShoreWare Director, gives the system administrator an overview of the ShoreTel system status. It includes information about each site and the corresponding switches, ports, servers, and service. This is the first place you should look to determine the status of the system.

6.6.2 Log File Size and Age

The Other Parameters window under Administration → System Parameters contains fields that show the maximum size and age of log files. These values must be set on a server, not in the V model Director's switch area. Figure 6-8 shows the default values for a V model switch (250 Mbytes and 3 days). In the current release, these log file values cannot be set here, only displayed.

Due to the constraints on CF memory space, log files should not be allowed to consume unnecessary amounts of memory, so after the V model switch has been added, change these values from a server.

6.6.2.1 Event Logs

The ShoreTel system uses the Windows Event Log, viewed using the Event Viewer, to report information and errors that are of interest to system administrators. You can use the event logs in conjunction with Quick Look to determine the overall health of the system. You can also use the event log to gather information about an event that is no longer a problem. For example, the event log may provide information about an overnight T1 outage that was corrected is no longer evident.

Each system task reports when it starts and stops. These messages can be helpful in determining whether the system has started correctly. Events, such as switches losing connection to the server or rebooting, are also reported.

6.6.2.2 System Logs

The ShoreTel system stores a number of engineering-level log files that record transaction information about every event in the ShoreTel system. The logs are used to help ShoreTel with debugging problems that may arise during system operation. In most cases, these logs are difficult to interpret and will require the assistance of ShoreTel Customer Support to understand.

6.6.3 Monitoring CF Memory Usage

A V model switch provides CF usage data to the ShoreWare Director and a CLI command. If the CF card becomes full, it cannot accept new voice mail. This section describes four Director windows that provide different views of memory usage on a CF card. Certain Director windows variously show the CF's used space, free space, percent of usage, and color-coded alerts. Also, the administrator can create an event filter to send notification of high memory usage.

Typical CF capacities are 1 Gbyte, 2 Gbytes, and 4 Gbytes, although the V model switch is designed to support much larger capacities. The proper planning of voice mail usage and aging can help prevent excessive buildup of voice mail on the CF card.

In the first of the windows that follow (Figure 6-16), a new column at the far right of the Maintenance - Quick Look shows the percent of disk usage for all servers. For a V model switch, this disk usage refers to the CF card. An administrator can immediately see how full the CF card is, but this screen does not provide the colored alarm indicator or the amounts of free and used space for the switch.

In Quick Look, an administrator can select a particular V model switch and see the configuration and usage numbers for individual voice mailboxes (Figure 6-17). In Quick Look, the colored alerts are yellow (1–10 messages left) and red (for mailbox is full). In other windows, these colors can mean other thresholds.

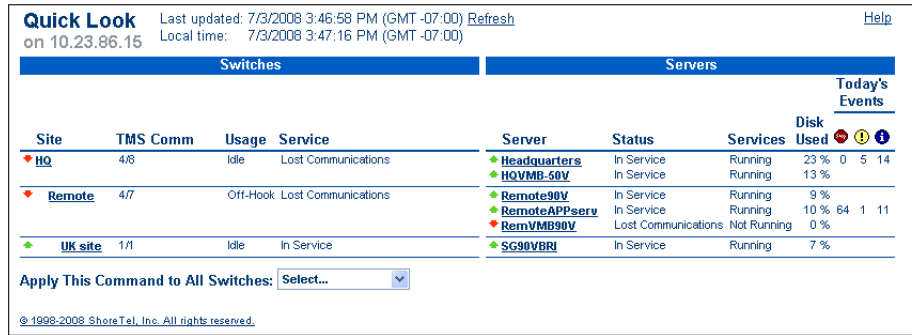


Figure 6-16 Percent of Disk Usage in the Quick Look Window

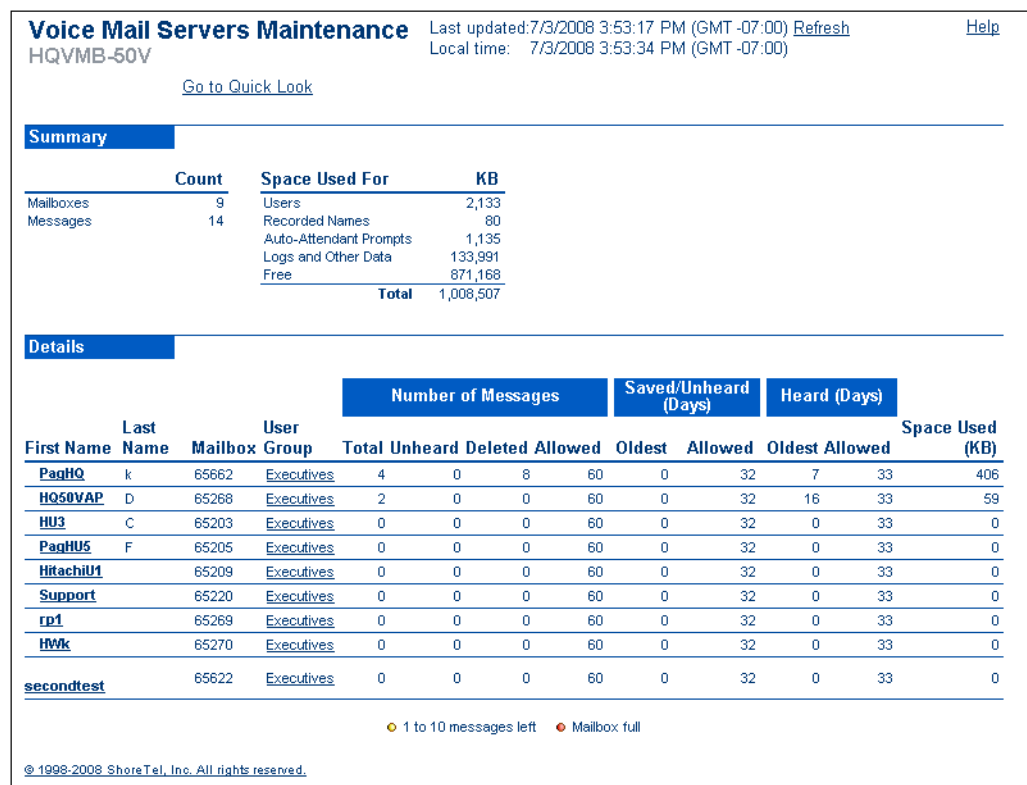


Figure 6-17 Specific Voice Mail Availability through the Quick Look Window

For the amounts of free and used disk space in Mbytes and any alerts, the administrator can go to the Voice Mail Servers Maintenance - Summary window. This window (Figure 6-18) shows the amount of free and used space in Mbytes for each voice mail server. For memory usage warnings, an alarm indicator appears in the Free Space (MB) column next a specific switch's value. In Figure 6-18, the small circle next to the HQVMB-50V Free Space value is a yellow alert. The legend at the bottom of the screen shows that yellow means CF free space is at 25-50% of capacity. Red means free space is at or below 25% of capacity.

The next degree of more granularity is information for each user. In Figure 6-19, the screen for voice mail server HQVMB-50V shows the current number of Kbytes of voice mail for each user and the configuration details for each user. The alerts in this window are yellow for 1–10 messages left and red for mailbox full.

ShoreWare Director displays the CF memory usage as a disk usage value.

Voice Mail Servers Maintenance Last updated: 7/3/2008 3:57:12 PM (GMT -07:00) [Refresh](#) [Help](#)
 Summary Local time: 7/3/2008 3:57:30 PM (GMT -07:00)

Voice Mail Servers						
Site	Voice Mail Server	Mailboxes	Messages	Space Used (MB)	Free Space (MB)	Last Successful Backup
HQ	◆ Headquarters	21	34	18266	59809	
HQ	◆ HQVMB-50V	9	14	134	850	07/03/2008 - 02:09:53 AM
Remote	◆ Remote90V	7	3	189	1736	None
Remote	◆ RemoteAPPserv	2	4	8492	69563	
Remote	◆ RemVMB90V	0	0	0	0	
UK site	◆ SG90VBRJ	3	1	144	1781	None

● 25 to 50% left ● 25% Left

© 1998-2008 ShoreTel, Inc. All rights reserved.

Figure 6-18 Disk Usage for Each Server in Voice Mail Servers Maintenance

Voice Mail Servers Maintenance Last updated: 2/6/2008 10:37:24 AM (GMT -08:00) [Refresh](#) [Help](#)
 HQVMB-50V Local time: 2/6/2008 10:37:34 AM (GMT -08:00)

[Go to Quick Look](#)

Summary			
	Count	Space Used For	KB
Mailboxes	8	Users	618
Messages	14	Recorded Names	115
		Auto-Attendant Prompts	133
		Logs and Other Data	625,129
		Free	382,512
		Total	1,008,507

Details											
First Name	Last Name	Mailbox	User Group	Number of Messages			Saved/Unheard (Days)		Heard (Days)		Space Used (KB)
				Total	Unheard	Allowed	Oldest	Allowed	Oldest	Allowed	
HU1	A	210	Executives	4	0	60	0	77	8	35	149
HU2	B	202	Executives	4	4	60	9	77	0	35	66
HQSIPIUSR		666	Executives	2	2	60	6	77	0	35	63
HU4	D	204	Executives	4	0	60	0	77	8	35	62
wg3		127	Executives	0	0	60	0	77	0	35	0
HQ60VAP		268	Executives	0	0	60	0	77	0	35	0
tp1		269	Executives	0	0	60	0	77	0	35	0
HVK		270	Executives	0	0	60	0	77	0	35	0

● 1 to 10 messages left ● Mailbox full

© 1998-2007 ShoreTel, Inc. All rights reserved.

Figure 6-19 CF Card Usage per User on HQVMB-50V

Configuring Trunks

This chapter provides instructions for configuring trunk groups and trunks from ShoreWare Director.

The sections included in this chapter are:

- Overview
- Trunks
- Individual Trunks

The expanded *Trunks* link in the navigation frame is shown in Figure 7-1.

7.1 Overview

Before you begin, you should understand the different trunk types and trunk features that are supported on the ShoreTel 6.1 system.

- A very thorough description of the types of trunks and their associated features is included in the *ShoreTel 8 Planning and Installation Guide*, Chapter 5, “Trunk Planning and Ordering.”
- A detailed description of how the dialing plan, network call routing, and digit manipulation operates is included in the *ShoreTel 8 Planning and Installation Guide*, Chapter 6, “Dialing Plan and Network Call Routing.”

For more information about the features supported outside the U.S. and Canada, refer to the *ShoreTel 8 Planning and Installation Guide*, Appendix A, “International Planning and Installation.”

For an overview of the various Trunk Types and Trunk Features, please refer to the “Trunk Planning and Ordering” chapter in the *ShoreTel 8 Planning and Installation Guide*.

7.2 Setting Up Trunks

Click *Trunk Groups* under the *Trunks* link in the navigation frame to invoke the *Trunk Groups* list page, as shown in Figure 7-1.

Name	Type	Site	Trunks	DID	Destination	Access Code
Analog Loop Start	Analog Loop Start	Headquarters	0	No	818-5777	9
ATT DID Incoming	Digital Wink Start	Headquarters	8	Yes	818-5777	9
ATT Pri1 Trk Grp	PRI	Headquarters	46	Yes	818-5777	9
ATT T1 2way E&M	Digital Wink Start	Headquarters	8	Yes	818-5777	0
bla T1	Digital Wink Start	Headquarters	0	Yes	818-5777	9
Digital Loop Start	Digital Loop Start	Headquarters	0	No	818-5777	9
Digital Wink Start	Digital Wink Start	Headquarters	8	No	818-5777	9

Figure 7-1 Trunk Groups List Page

The columns in the *Trunk Groups* list page follow:

- **Name:** This is the name of an existing trunk group. Clicking a name invokes the *Trunk Group* edit page, where you can edit the trunk group configuration.
- **Type:** This is the type of trunk group.
- **Site:** This is the site of the trunk group. Note that a trunk group cannot span sites. For information about configuring sites, refer to Chapter 3, starting on page 39.
- **Trunks:** This is the number of trunks in the trunk group.
- **DID:** This column shows whether the trunk supports DID. The trunk group supports at least one of the following:
 - DID
 - DNIS
 - Extension
- **Destination:** This is the destination where incoming calls are routed. This may be the auto-attendant, the operator, or an individual user's extension. According to the options you have set, incoming calls are routed, in the order they are listed, to:
 - DNIS
 - DID
 - Extension
 - Destination

Destination is always available as the last choice for routing.
- **Access Code:** This displays the access code for the trunk group.

7.2.1 Add or Edit a Trunk Group

To add a new trunk group, on the *Trunk Groups* list page, specify a site from the *Add new Trunk Group at site* drop-down list, select a trunk group type from the *of type* drop-down list, and click *Go*. To edit an existing trunk group, click a trunk group from the list of trunk groups in the *Name* column on the *Trunk Groups* list page.

When you add a new trunk group or click the name of an existing trunk group from the *Trunk Groups* list page, the appropriate *Trunk Group* edit page appears. Figure 7-2 shows the *T1 PRI Trunk Group* edit page, which is a superset of all the other *Trunk Group* edit pages except the *E1 Trunk Group* edit page and the *SIP Trunk Group* edit page.

The parameters are defined as follows:

- **Name:** This is the name of the new or existing trunk group.
- **Site:** The Name of the site of the trunk group is displayed.
- **Language:** Select the language for the trunk group from the drop-down list.
- **Teleworkers:** This parameter only appears when editing SIP trunk groups.

When this check box is enabled, third-party SIP devices will use the public network for transmitting audio on intersite calls, while the signaling will use intersite bandwidth. If not selected, the call audio will use intersite bandwidth.

- **Enable SIP Info for G.711 DTMF Signaling:** This parameter only appears when editing SIP trunk groups.

Trunk Groups
Edit SIP Trunk Group

[New](#)
[Copy](#)
[Save](#)
[Delete](#)
[Reset](#)
[Help](#)

* modified

Edit this record [Refresh this page](#)

Name:

Site: Headquarters

Language:

Teleworkers

Enable SIP Info for G.711 DTMF Signaling

Enable Digest Authentication

User ID:

Password:

Inbound:

Number of Digits from CO:

DNIS [Edit DNIS Map](#)

DID [Edit DID Range](#)

Extension

Translation Table:

Prepend Dial In Prefix:

Use Site Extension Prefix

Tandem Trunking

User Group:

Prepend Dial In Prefix:

Destination: [Search](#)

Outbound:

Network Call Routing:

Access Code:

Local Area Code:

Additional Local Area Codes: [Edit](#)

Nearby Area Codes: [Edit](#)

Trunk Services:

Local

Long Distance

International

n11 (e.g. 411, 611, except 911 which is specified below)

911

Easy Recognizable Codes (ERC) (e.g. 800, 888, 900)

Explicit Carrier Selection (e.g. 1010xxx)

Operator Assisted (e.g. 0+)

Caller ID not blocked by default

Trunk Digit Manipulation:

Remove leading 1 from 1+10D
Hint: Required for some long distance service providers.

Remove leading 1 for Local Area Codes (for all prefixes unless a specific local prefix list is provided below)
Hint: Required for some local service providers with overlay area codes.

Dial 7 digits for Local Area Code (for all prefixes unless a specific local prefix list is provided below)
Hint: Local prefixes required for some local service providers with mixed 7D and 1+10D in the same home area.

Local Prefixes: [Go to Local Prefixes List](#)

Prepend Dial Out Prefix:

Off System Extensions: [Edit](#)

Translation Table:

Figure 7-2 Trunk Group Edit Page

Enable this check box when using third-party SIP devices that do not support DTMF negotiation via RFC 2833. If the device does not support RFC 2833 and this check box is not enabled, DTMF negotiation will fail.

- **Enable Digest Authentication:** This parameter only appears when editing SIP trunk groups.

Enable this check box when using dynamic SIP trunks to provide enhanced security via User ID and Password authentication.

- *User ID*—This is the user ID associated with the third-party SIP device.
- *Password*—This is the password associated with the third-party SIP device.

Inbound Settings

The **Inbound** settings let you route inbound calls to a single destination, or route calls to a specific destination using DID and DNIS digits.

- **Number of Digits from the CO:** This specifies the maximum number of digits expected from the central office (for User PRI configured trunks). Digit collection terminates when the maximum number of digits are received, when the digit collection time-out is reached, or when an exact match is found.
Network PRI trunks connected to legacy PBXs collect digits from the legacy PBX side. When the ShoreTel 6.1 system detects a trunk access code, it ignores the *Number of Digits from the CO* parameter and routes the call according to the dialing plan.
- **DNIS:** When DNIS is checked, click **Edit DNIS Map** to add or delete entries in the DNIS Map.
- **DID:** When DID is checked, click **Edit DID Range** to add or edit the DID Range as well as view the DID Digit Map.
- **Extension:** If you check this option, calls will route directly to the extension based on the digits received from the central office without any additional configuration. This is very useful when configuring a tie trunk connected to a legacy PBX. Note that the extension length must match the number of digits from the CO.
 - *Translation Table*—When using the On-Net Dialing feature, this parameter allows you to specify a digit translation table that is used to strip one or more digits from calls between two systems with extensions of different lengths.
 - *Prepend Dial In Prefix*—When using the On-Net Dialing feature, this parameter allows you to add one or more digits to calls between two systems with extensions of different lengths.
 - *Use Site Extension Prefix*—When using the On-Net Dialing feature, this parameter allows you to specify a site extension prefix that will be added to calls from a system that does not have a prefix to another system that does have a prefix.
- **Tandem Trunking:** Tandem trunking allows legacy voice systems to utilize a ShoreTel 6.1 system for outbound dialing. The ShoreTel 6.1 system supports network-side PRI, allowing ShoreTel 6.1 systems to flexibly support digital tie trunks to other systems.
 - *User Group*—Tandem calls are associated with a user group for outbound trunk selection. Inbound calls that are recognized as tandem calls are then redirected to an outbound trunk based on the call permissions and trunk group access associated with the user group set in Director.
 - *Prepend Dial In Prefix*—When needed, you can specify a “dial in prefix” which is pre-pended to digits collected on tandem calls. The concatenated set of digits is then be used in outbound trunk selection for the tandem call.

- **Destination:** You must enter a destination. All inbound calls are routed to a specific extension (user, workgroup, route point) or to a specific menu. If you selected other destination options, this is the last choice destination.

Inbound calls first try to match a DNIS entry, then a DID entry, followed by an Extension entry, and finally Tandem Trunking. If no match is found, the inbound call will be routed to the destination you set. If you create a trunk group, the destination will be the default auto-attendant.

An individual trunk group cannot have the overlapping DID or DNIS numbers (received digits).

Users, Menus, Workgroups, and Route Points can have one DID number but multiple DNIS entries.

Outbound Settings

- **Network Call Routing:** When enabled using the *Outbound* check box, this lets you configure the following trunk group *Network Call Routing* options:
 - *Access Code*—Enter the appropriate trunk access code for this trunk group. This is typically “9” in the U.S. and Canada.
 - *Local Area Code*—Enter the local area code for this trunk group. This area code is used for Network Call Routing as well as Digit Manipulation.
 - *Additional Local Area Codes*—Click *Edit* to enter any additional area codes that are typically associated with overlay area codes. These additional local area codes are used for Network Call Routing as well as Digit Manipulation.
 - *Nearby Area Codes*—Click *Edit* to enter area codes that are considered nearby, or “free”, for this trunk group. The nearby area codes are used for Network Call Routing.
- **Trunk Services:** This parameter lets you configure the trunk group for the outbound services described below. Unless specified, if an option is not enabled, the trunk group does not provide that service.
 - *Local*—Select this check box to enable local calls.
 - *Long Distance*—Select this check box to enable long-distance calls.
 - *National Mobile (not shown)*— This check box only appears for PRI trunks in countries with “caller pays” billing plans (e.g. Ireland). Select this check box to allow users to call mobile numbers. Clear the checkbox if you do not want to incur the associated costs of allowing users to call mobile phones in “caller pays” environments.
 - *International*—Select this check box to enable international calls.
 - *n11 (e.g., 411 or 611, but not 911, which is specified below)*—Select this check box to enable telephone service calls, such as directory assistance.
 - *911*—Select this check box to enable emergency 911 calls.
 - You must have at least one trunk group per site that allows 911 calls.
 - *Easy Recognizable Codes (ERC) (e.g., 800, 888, 900)*—Click this check box to enable services such as toll-free dialing.
 - *Explicit Carrier Selection (e.g., 1010xxx)*—Click this check box to enable dialing special numbers that let the caller select a long-distance carrier.

- *Operator Assisted (e.g., 0+)*—Click this check box to enable the trunk group to dial the operator.
- *Called ID not blocked by default*—Click this check box to pass Caller ID information by default on outbound calls. To block all calls, clear this option. Note that in the United States, the user can override this option with Vertical Service Codes.
- **Trunk Digit Manipulation:** This parameter lets you control how the trunk group manipulates the telephone number before outpulsing the digits to the central office. All North American numbers are converted into 1+10-digit format internally before being passed to the trunk group for digit manipulation.
 - *Remove leading 1 from 1+10D*—Click this check box to drop the leading “1”. Dialing only ten digits is required by some long-distance service providers.
 - *Remove leading 1 for Local Area Codes (for all prefixes unless a specific local prefix list is provided below)*—Click this check box to drop the leading “1” for the local area codes (Local and Additional Local). Dialing only ten digits for local area codes, particularly with overlay area codes, is required by some local service providers. If a local prefix list provided, the leading “1” is removed for the all entries in the list.
 - *Dial 7 digits for Local Area Code (for all prefixes unless a specific local prefix list is provided below)*—Click this check box to enable the trunk to dial local numbers in the local area code with seven digits. This is required by some local service providers.

If a local prefix list provided, seven digits are dialed for all entries in the list (Local Area Code only, not Additional Local Area Codes).

- **Local Prefixes:** Click the *Go to Local Prefixes List* link to view, add, 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.

You can import local prefixes from a CSV file. For more information, see Section 7.2.2.

- **Prepend this Dial Out Prefix:** Select a prefix from the drop-down list. It will be 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. The Dial Out Prefix is not applied to Off-System Extension calls.
- **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. See also Section 12.3.1 on page 277.

The Dial Out Prefix and Digit Manipulation rules are not applied to Off-System Extensions.

- **Translation Table:** Specify the digit translation table that is used to strip one or more digits from calls between two systems with extensions of different lengths.

ShoreTel 6.1 does not apply inbound digit treatment to digit strings beginning with a trunk access code (such as 9) as would occur in tie trunk configurations. Digit strings beginning with a trunk access code are routed according to the dial plan.

A user group named Account Codes Service is created for use by the Account Codes Service. It is not available for assignment to users but it may be edited.

7.2.2 Importing Local Prefixes

You can import or export local prefixes in CSV format. Local prefix lists can be purchased or obtained free from various web sites.

To import local prefixes from a CSV file:

Step 1 Click *Go to Local Prefixes List* in the Trunk Groups panel. If no local prefix lists are defined, the *Local Prefixes* edit panel appears as shown in Figure 7-3.

Figure 7-3 Local Prefixes Edit Page

If a local prefix list already exists, the *Local Prefix* list page appears as shown in Figure 7-4. Click *Add New List*.

Figure 7-4 Local Prefixes List Page

Step 2 Click *Import*. The *Local Prefixes Import* dialog box appears as shown in Figure 7-5

Figure 7-5 Local Prefixes Import Dialog Box

Step 3 Enter the path and name of the file you want to import or click *Browse* to search for the file.

Step 4 Click *Upload*. The *Local Prefixes* edit page appears as shown below.

Step 5 Edit the list as needed. You can re-name the list and add, edit, or remove prefixes.

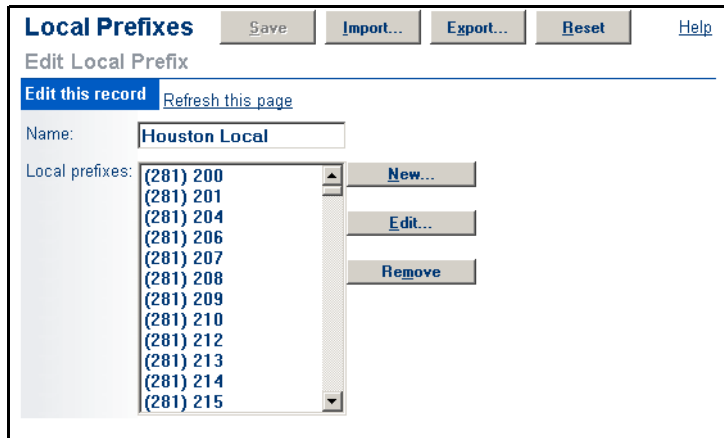


Figure 7-6 Local Prefixes Edit Page, Populated

Step 6 Click Save. The list you created is now available from the *Local Prefix* drop-down list.

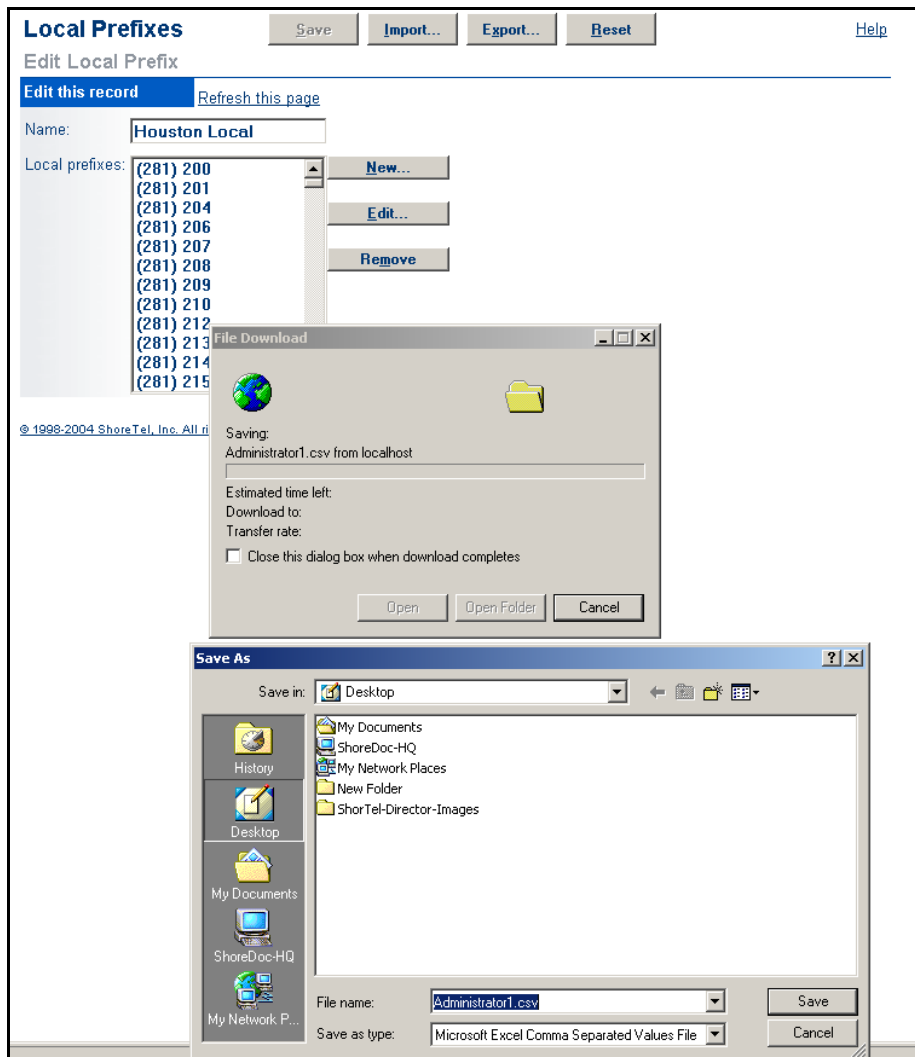


Figure 7-7 Saving the CSV File

To export a prefix list:

- Step 1 Click *Go to Local Prefixes List* from the Trunk edit page. If a local prefix lists already exist, the *Local Prefix* list page appears as shown in Figure 7-4.
- Step 2 Click the list you want to export. The *Local Prefixes* edit page appears as shown in Figure 7-3.
- Step 3 Click *Export*. Your web browser opens the file in a new window. Select *File -> Save As* and save the web page as a TXT file (see Figure 7-7).
- Step 4 Click *Save*.

7.2.3 Edit DID Range

To set a DID range, click *Edit DID Range* from the appropriate *Edit Trunk Group* page to invoke the *DID Range* page (Figure 7-8).

Figure 7-8 DID Range Page

For each block of DID numbers, enter the base phone number and the number of phone numbers supplied by that block in the appropriate fields, and click *Add this record*. When finished, click *Save*. You can configure multiple ranges per trunk group.

To view the currently configured DID Digit Map (Figure 7-9), click *View DID Digit Map*. From here, you can view pertinent information about each DID number in the ShoreTel 6.1 system.

DID Number	Received Digits	Destination
+1 (213) 333-0001	0001	Adam Ant
+1 (213) 333-0002	0002	Donald Duck
+1 (213) 333-0003	0003	Johnathan Seagull
+1 (213) 333-0004	0004	Ernie Els
+1 (213) 333-0005	0005	Greta Garbo

Figure 7-9 DID Digit Map

7.2.4 Edit DNIS Digit Map

To edit the DNIS Digit Map, click *Edit DNIS Map* from the appropriate *Trunk Group* edit page to invoke the *DNIS Digit Map* edit page (Figure 7-10).

Figure 7-10 DNIS Digit Map Edit Page

For each DNIS entry, enter the digits you expect from the service provider when this number is dialed in the *Received Digits* field. Then, type a meaningful name into the *Dialed Number* field. This can be a mix of digits and alphabetic characters. This friendly description will be presented in the Personal Call Manager and CDR Reports.

Select a *Destination* extension using the *Search* button and then click *Add this record*.

First Name	Last Name	Extension
ip6		4181
ip7		4182
ip8		4183
ip9		4184
Kwang	Lower	4114
MAFuser		4705
New Bridged Call Appearance		4130
New Hunt Group		4217
New Multiple Extension Appearance		4500
New Workgroup		4219
patrick	visitacion	4119
Priya		4175
Robert	laFrentz	4315
Sindhu		4503

Figure 7-11 Selecting a Destination

You can also select an off-system extension to which calls will be routed. First select an extension range and then enter the extension in the text box.

When finished, click *Save*.

7.2.5 Additional Local Area Codes

To configure additional local area codes, click the associated button on the *Trunk Group* edit page (Figure 7-2). The *Addition Local Area Codes* dialog box appears (Figure 7-11).

Click *New* to add a new additional local area code and *Remove* to delete one.

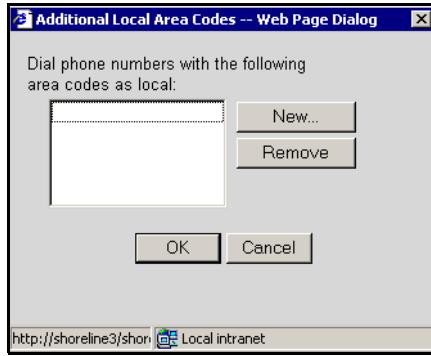


Figure 7-12 Additional Local Area Codes Dialog

7.2.6 Nearby Area Codes

To configure nearby area codes, click the associated button on the *Trunk Group* edit page (see Figure 7-2). The *Nearby Area Codes* dialog box appears (see Figure 7-13).

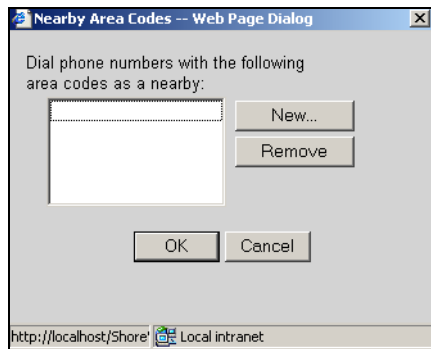


Figure 7-13 Nearby Area Codes Dialog

Click *New* to add a nearby area code and *Remove* to delete one.

7.2.7 Prefix Exceptions

To configure the local prefixes list, click the link *Go to Local Prefixes List* located near the bottom of the *Trunk Group* edit page (see Figure 7-2). The Local Prefixes dialog box is presented.

If you click *Delete*, changes take place immediately.

To configure local prefixes, click *add new list* or an existing list to bring up the Local Prefixes dialog box (see Figure 7-14).

Changes take place when you click *OK* within the Local Prefixes dialog.

7.2.8 Off-System Extensions

If you are using off-system extensions, you can list them by clicking the *Edit* button on the *Trunk Group* edit page. Off-system extensions are typically used when setting up a tie trunk to a legacy PBX and configuring coordinated extension dialing. Also refer to Section 12.3.1 on page 277. Figure 7-15 shows the dialog boxes used to define ranges of extensions.

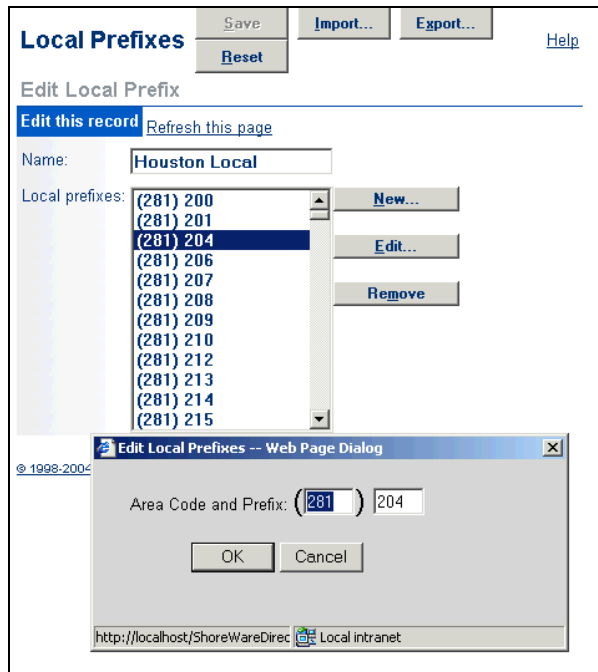


Figure 7-14 Local Prefixes Dialog Box

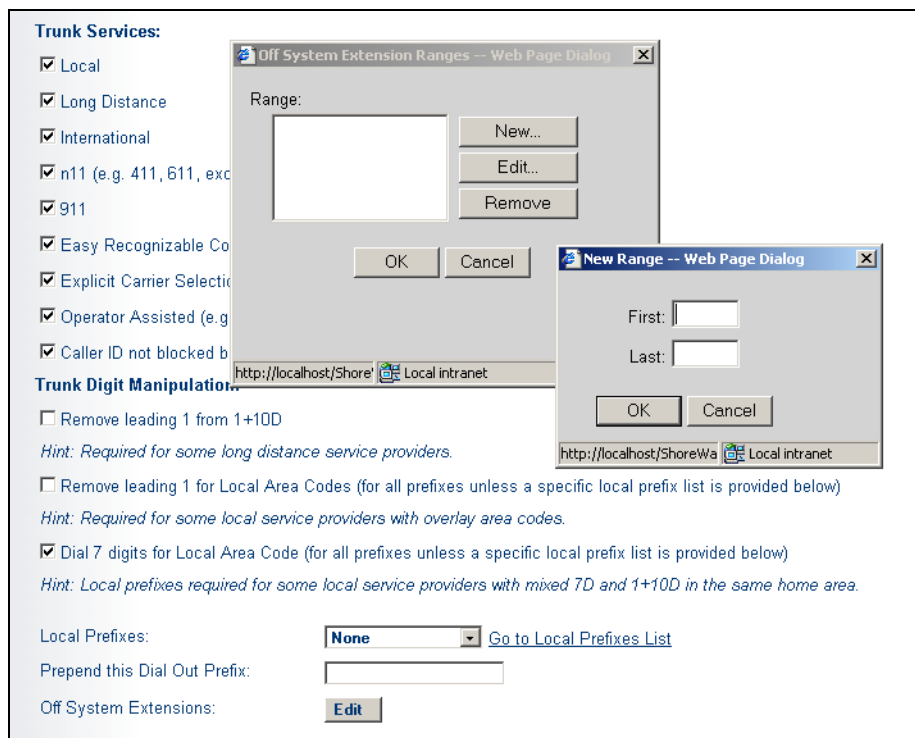


Figure 7-15 Configuring Off-System Extensions

7.2.9 Tandem Trunking

Tandem trunking treats digits on an incoming trunk call as a PSTN number. Received digits are tested against DNIS, DID, Extension, and Tandem Trunking, in that order. When Tandem Trunking is enabled, the number of digits from the CO may have no effect if the first digit(s) matches a Trunk Access Code. To define trunk access and call permissions, associate a user group with the tandem trunk group.

Any Dial In Prefix is pre-pended to each set of inbound digits. You can use DNIS/DID/Extension matching with a Dial In Prefix.

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

Tandem calls are reported in the Trunk Detail and Trunk Summary reports, with incoming and outgoing legs reported according to the reports' formats.

7.2.10 Centrex Flash

Centrex Flash can be programmed on one of the custom buttons, allowing users to transfer the call to another number. When the custom button is pressed, a flash is generated on the current call, dial tone is presented by the central office (CO), and the user can dial any PSTN number. When ring-back is heard, the handset can be hung up to complete the transfer.

Note that the user will be connected directly to the central office, so there will be no access code required, no permissions checking, no account code support, and no CDR logging of the second call.

Centrex Flash is useful in branch offices or small office environments with a limited number of analog Centrex lines. If an external caller needs to be transferred to an external number, the two trunks will be cleared, instead of quickly busying out the trunks after a few transfers. In this manner, the feature reduces the number of physical trunks needed to transfer calls, since no trunks are in use after the transfer is completed.

Centrex transfer is supported only on analog loop-start trunks. If the call is not on an analog loop-start trunk, the operation will have no effect. The trunk on which the call exists must be configured on one of the following switches:

- ShoreGear 40/8
- ShoreGear 60/12
- ShoreGear 120/24
- ShoreGear-E1
- ShoreGear-T1

The feature replaces a trunk-to-trunk transfer in which two trunks are tied up for the duration of the call. The current call must be connected and also must be a two-party call.

For information on configuring Centrex Flash, please see Section 10.4.8 on page 207.

7.2.11 Pause in Dialing – Trunk Access

The Pause in Dialing – Trunk Access feature allows the insertion of commas (,) into a trunk group's Prepend Dial Out prefix to specify one-second pause periods during the transmission of pulse digits. Pause periods are permitted in the following trunk group types that send digits as pulses:

- Analog Loop Start
- Digital Loop Start
- Digital Wink Start

Example: Assume the prepend dial out prefix of an analog trunk group is “9,,”. When a user dials “914085551111”, the following pulse-digit sequence is transmitted on the trunk: “9<silence for two seconds>14085551111”.

Commas are permitted anywhere within the Prepend Dial Out Prefix.

Feature restrictions include:

- The pause cannot be used to insert account codes.
- Commas in Call Manager dial strings are ignored,.
- Commas in dial strings sent by other TAPI applications are ignored.

To enter a dial pause for a specific trunk group, perform the following:

Step 1 Open the Trunk Groups List window by selecting *Administration -> Trunks -> Trunk Groups* from the Director Main Menu.

Step 2 Open the Trunk Groups Edit panel, as shown in Figure 7-16, for the desired trunk group by clicking the name of that trunk group.

You can also open this panel by creating a trunk group.

The screenshot shows the 'Trunk Groups' edit panel for 'Digital Wink Start'. The panel includes the following elements:

- Buttons:** New, Copy, Save, Delete, Reset, Help.
- Form Fields:**
 - Name: Digital Wink Start
 - Site: NJ
 - Language: English(US)
- Trunk Digit Manipulation:**
 - Remove leading 1 from 1+10D
Hint: Required for some long distance service providers.
 - Remove leading 1 for Local Area Codes (for all prefixes unless a specific local prefix list is provided below)
Hint: Required for some local service providers with overlay area codes.
 - Dial 7 digits for Local Area Code (for all prefixes unless a specific local prefix list is provided below)
Hint: Local prefixes required for some local service providers with mixed TD and 1+10D in the same home area.
- Local Prefixes:** None (dropdown), [Go to Local Prefixes List](#)
- Prepend Dial Out Prefix:** (text input field, highlighted with a red arrow)
- Off System Extensions:** Edit
- Translation Table:** <None> (dropdown)

Figure 7-16 Modifying the Prepend Dial Out Prefix setting

Step 3 Enter the code in the Prepend Dial Out Prefix data entry field located near the bottom of the panel.

Step 4 Press the Save button at the top of the panel.

7.3 Individual Trunks

This section explains how to configure individual trunks once you have created the associated trunk groups as described in Section 7.2.

To invoke the *Trunks by Group* page, click *Trunks* from the navigation frame, and then click *Individual Trunks*. The *Trunks by Group* page appears, as shown in Figure 7-17.

Name	Group	Type	Site	Switch	Port/Channel	SIP IP Address
blah	Pat to Swarna Tie TG	PRI	HeadQuarters	FujiT1 Tie Test Patrick to Swarna	1	
blah (10)	Pat to Swarna Tie TG	PRI	HeadQuarters	FujiT1 Tie Test Patrick to Swarna	10	
blah (11)	Pat to Swarna Tie TG	PRI	HeadQuarters	FujiT1 Tie Test Patrick to Swarna	11	
blah (12)	Pat to Swarna Tie TG	PRI	HeadQuarters	FujiT1 Tie Test Patrick to Swarna	12	
blah (13)	Pat to Swarna Tie TG	PRI	HeadQuarters	FujiT1 Tie Test Patrick to Swarna	13	
blah (14)	Pat to Swarna Tie TG	PRI	HeadQuarters	FujiT1 Tie Test Patrick to Swarna	14	
blah (15)	Pat to Swarna Tie TG	PRI	HeadQuarters	FujiT1 Tie Test Patrick to Swarna	15	
blah (16)	Pat to Swarna Tie TG	PRI	HeadQuarters	FujiT1 Tie Test Patrick to Swarna	16	

Figure 7-17 Trunks by Group Page

To select a trunk group, select a site from the *Add new trunk at site* drop-down list, then select a trunk group from the *In trunk group* drop-down list, and click *Go*. The trunks belonging to the selected trunk group appear. The columns in the *Trunks by Group* page are as follows:

- **Name:** This is the name of an individual trunk in the group. Click an entry in the *Name* column to invoke the *Trunks* edit page to edit the individual trunk's parameters.
- **Group:** This is the trunk group name. Click a group name to invoke the *Trunk Group* edit page and edit the trunk group's parameters.
- **Type:** This ID can range from 1 - 6 and corresponds to the trunk type (e.g. analog DID, analog loop start, SIP, etc.)
- **Site:** This is the location of the trunk and can be the Headquarters location or one of the remote locations.
- **Switch:** This is the IP host name of the ShoreGear voice switch to which the individual trunk is connected.
- **Port/Channel:** This is the port number or channel to which the individual trunk is connected.
- **SIP IP Address:** This IP address only applies to SIP trunks and corresponds to the SIP end point device.

7.3.1 Add or Edit a Trunk

Whether you choose to add a new trunk or edit an existing trunk, the *Trunks* edit page shown in Figure 7-18 appears.

7.3.1.1 Non SIP Trunk Parameters

The parameters on the *Trunks* edit page (for non-SIP trunk types) are as follows:

- **Site:** This is the name of the site at which the trunk and trunk group are located.

Figure 7-18 Trunks Edit Page

- **Trunk Group:** This is the name of the trunk group to which the trunk group belongs. You cannot change the name of the trunk group on this page.
- **Name:** This text-entry field lets you enter the name of the individual trunk.
- **Switch Channel:** This drop-down list select the channel to which this trunk connects.
- **Jack #:** This is the patch-panel jack number that is associated with the trunk's switch port. This is an optional parameter.

Figure 7-19 SIP Trunks Edit Page

7.3.1.2 SIP Trunk Parameters

The parameters on the *Trunks* edit page (for SIP trunks) are as follows:

- **Site:** This is the name of the site at which the trunk and trunk group are located.
- **Trunk Group:** This is the name of the trunk group to which the trunk group belongs. You cannot change the name of the trunk group on this page.
- **Name:** This text-entry field lets you enter the name of the individual trunk.
- **Switch:** This drop-down list lets you select the switch to which this trunk is connected.
- **SIP Trunk Type:** Use the radio button to assign a static IP address or have one dynamically assigned.

Number of Trunks (1 - 120): This is the number of trunks in the trunk group. This field only appears when a SIP trunk is first created. When the SIP trunk is subsequently modified, this field does not appear.

Configuring IP Phones

This chapter explains how to:

- Set IP address ranges
- Set IP phone options
- Add an IP phone password
- Assign configuration switches to support the IP phones
- Add IP phone users to the system
- Select a user group for unassigned IP phones
- Manage the IP phones

8.1 Overview

ShoreTel 6.1 supports IP phones connected through ShoreGear 120/24, Shoregear 60/12, and ShoreGear 40/8 voice switches. Each ShoreGear 120/24 voice switch supports up to 120 IP phones, the ShoreGear 60/12 supports up to 60 IP phones, and the ShoreGear 40/8 supports up to 40 IP phones.

Before you can add IP phone users through ShoreWare Director, you must first:

- Set the boot parameters in the individual IP phones if you are using static IP addresses. For more information, see your *ShoreTel 8 Planning and Installation Guide*.
- Add and configure all ShoreGear-120/24, ShoreGear-60/12, and ShoreGear-40-8 voice switches that will be supporting IP phones. For information on allocating switch ports to IP phone support, see Chapter 5, starting on page 55.

For more information on these steps, see the instructions given in the *ShoreTel 8 Planning and Installation Guide*. When you have completed the installation process, the ShoreTel 6.1 system automatically detects IP phones connected to the network.

Click the **IP Phones** link in the navigation frame to expand and display additional links, as shown in Figure 8-1.

8.2 Setting IP Address Ranges

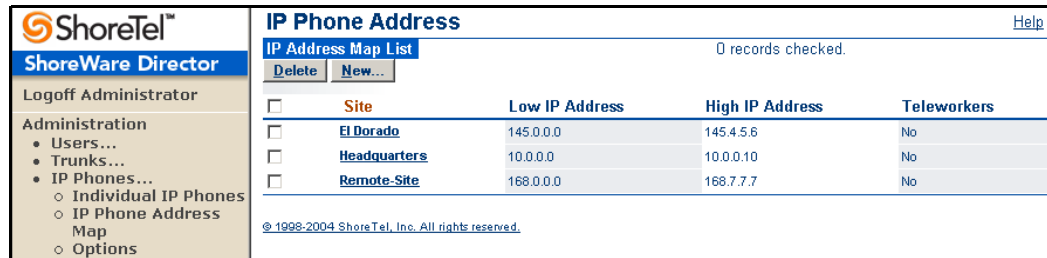
If your system consists of more than one site (Headquarters), you must define an IP address range for IP phones at each site in the system. Setting ranges for each site ensures that new phones added to the system are associated with the correct site.

The ShoreTel Converged Conference Bridge uses IP phone ports to support conference call ports. Include the capacity of any ShoreTel Converged Conference Bridges in your IP address range calculations.

You can review all the IP phones in the system from the *IP Phone List* page (see Section 8.6.1 on page 117). You can also view IP phones by the switch supporting them from the *IP Phone Maintenance* page. For more information, see Chapter 18, starting on page 345.

8.3 IP Phone Address Map

To view the IP address range for each site, expand the *IP Phones* link and click *IP Phones Address Map*. The *IP Address Map List* page, shown in Figure 8-2, appears. The page lists the sites and associated IP address ranges.



<input type="checkbox"/>	Site	Low IP Address	High IP Address	Teleworkers
<input type="checkbox"/>	El Dorado	145.0.0.0	145.4.5.6	No
<input type="checkbox"/>	Headquarters	10.0.0.0	10.0.0.10	No
<input type="checkbox"/>	Remote-Site	168.0.0.0	168.7.7.7	No

Figure 8-1 IP Address Map List Page

To add a new range of IP phones, click *New* and enter the information in the *IP Phone Address Map Info* edit page (see Figure 8-2). To delete a range from the list, click the check box to the left of the site and click *Delete*.

8.3.1 Adding Teleworking IP Phones

Improved support for teleworking IP phones no longer requires a dedicated site and ShoreGear switch for the best codec selection and correct bandwidth management. IP phones that fall into the IP address ranges for any teleworking IP phones operate just as other IP phones with a few differences:

- The inter-site voice codec will be used for all calls to and from teleworking IP phones.
- For bandwidth management purposes, the teleworking IP phone is not associated with any site and calls to and from that phone will not deduct from the available bandwidth of the call agent switch for that phone.
- The IP phone will be hosted by an available call agent switch at the site associated with the IP address map. Numbers dialed from teleworking IP phones will be interpreted in the context of the dial plan for the site associated with the IP address map.

To edit the IP address range for a site:

Step 1 From the *Site* column on the IP Address Map List page, click the site for which you are setting a range. The *Site IP Address Range* edit page appears as shown in Figure 8-2.

Step 2 If you are setting the IP address range for a site other than the one shown in the *Site* drop-down list, select it from the list.

Step 3 Enter the lowest IP address in the *Low IP Address* text-entry field.

Step 4 Enter the highest IP address in the *High IP Address* text-entry field.

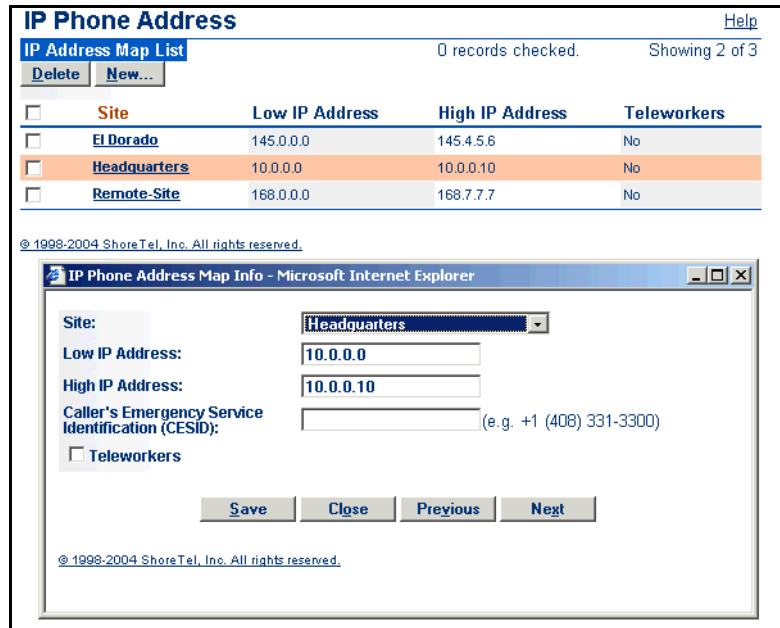


Figure 8-2 Site IP Address Range Edit Page

Step 5 Enter the Caller ID number in the Caller's Emergency Service Identification (CESID) field that you would like to assign to the IP phones within this address range. The CESID number will be sent to the service provider whenever a 911 call is placed from one of these IP phones. See Appendix B, starting on page 399, for more information.

Step 6 If you are setting a range for teleworking IP phones, click the *Teleworkers* check box.

Step 7 To set the new range, click *Save*. You can set ranges for other sites in the system by clicking *Previous* or *Next*.

8.4 Setting IP Phone Options

To set IP phone options such as passwords and configuration switches, click the *Options* link under the IP Phones link in the navigation frame. Figure 8-3 shows the *IP Phone Options* edit page that appears.

The parameters on the IP Phone Options edit page are defined as follows:

- IP Phone Configuration Switch 1 and IP Phone Configuration Switch 2:** You need to designate the switch that serves as the phone's Call Manager. This switch must be managed by the headquarters server. (See Section 5.5.1 on page 62 for more information). The IP addresses of these switches are downloaded to the IP phones whenever the IP phones are booted. These switches communicate with the ShoreTel server to determine which switch manages calls for a particular IP phone. You have the option of assigning two switches to this function, in case one switch fails. Every installation with IP phones must have at least one configuration switch.

To assign configuration switches, simply select an available switch from the drop-down lists for configuration switches 1 and 2. In a new installation, the ShoreTel 6.1 system

Figure 8-3 IP Phones Options Edit Page

automatically assigns the first two ShoreGear-120/24, ShoreGear-60/12, or ShoreGear-40/8 switches you configure as the configuration switches.

- **User Group for Unassigned Phones:** Unassigned IP phones are available for users configured for Any IP Phone. From the drop-down list, select the user group that has the call permissions you want unassigned IP phones to have.
- **IP Phone Announcement:** Enter a message of the day that displays on all IP phones except the ShorePhone IP210 phones. The text can be up to 19 characters long and appears left-justified on the phone display. If you wish to center the message, add leading spaces.
- **IP Phone Password:** This field is used only with ShorePhones that require a password. It is initially set to “1234”. It may be blank or be 1 through 8 digits long. It sets the default password for new IP phones coming online.
- **Enable IP Phone Failover:** When this box is checked, IP phones send a keep-alive message to their Call Manager switch every four minutes. If a response is not received, the IP phone attempts to contact an alternate Call Manager.

Changing the state of this field requires a reboot of all IP Phones. The process can take several minutes. Phones in the process of rebooting may drop calls.

- **Delay After Collecting Digits:** This field represents the timeout period for transferring calls. Instead of having to press a soft key to initiate a call transfer, the desired operation will occur automatically at the expiration of a configurable timeout period. Once all of the necessary digits have been entered (which could vary based on the site's dialing plan), digit collection stops and the timeout period begins counting down. At the end of the countdown, which can be as short as one second, the call is transferred.

The following features will also be affected by this change:

- Conference
- Dialing from the Directory
- Intercom
- On-hook dialing
- Park
- Pickup
- Redial
- Transfer
- UnPark

The timeout period can only be set once for the entire system. You cannot configure different timeout periods for different features or for different users. The timeout period cannot be configured via PCM or the IP phone interface.

Note that this automatic transfer behavior applies only to blind transfers. If a consultative transfer is placed, the call will remain in the call stack until the far end answers.

8.5 Adding Individual IP Phones Users

There are two ways to add IP phone users to the system. The first method requires you to assign IP phones to each user, while the second uses the Any IP Phone feature to allow users to assign their own phone from their desktop and voice mail. Using the Any IP Phone method simplifies the setup of multiple new users. You can use either or both methods, depending on your specific installation.

8.5.1 Adding Users with Any IP Phone

To add IP phone users using Any IP Phone:

Step 1 Click *Users > Individual Users* from the navigation frame.

Step 2 Click *Go*. The *Edit User* page appears, as shown in Figure 8-4.

Step 3 Enter the information for the first user you wish to add.

For information on user settings, see Chapter 10, starting on page 173.

Step 4 From *Home Port*, click *IP Phones* and select *Any IP Phone* from the drop-down list.

Step 5 From *User Group*, select a user group that allows extension reassignment. For more information, see Section 10.2.1 on page 175.

Step 6 When you have finished configuring the user, click *Save*.

Step 7 To add more users, click *Copy* and repeat Step 3 through Step 7 until you have added all the new users.

Step 8 Instruct the users to log in to their voice mail from their desktop and follow the prompts. Each user's extension will automatically be assigned to the IP phone he or she is using. Make sure that users know which phone they should initially use so that phones are not accidentally assigned to the wrong users.

All IP Phones are assigned to the "Headquarters" site. When you assign a specific IP phone, the user belongs to the site where the IP phone is located. Only IP phones associated with the currently selected site appear in the IP phone drop-down list.

After users have logged into voice mail to assign their IP phones, you can view the IP phones and users from the *IP Phone* list page (see Figure 8-5) or the *IP Phone Maintenance* page (see Figure 18-11 on page 356). For information about editing user information, see Section 10.4 on page 187. For in depth information about Office Anywhere, refer to Appendix A, starting on page 391.

Step 9 From *Add New User at Site*, select the site for the new users. Edit User Page

Users

Edit User

[Help](#)

New
Copy
Save
Delete
Reset

[Refresh this page](#)

▼ General
▶ Personal Options
▶ Distribution Lists
▶ Workgroups

First Name:

Last Name:

Number:

License Type: Extension and Mailbox ▼

Caller ID: (e.g. +1 (408) 331-3300)

DID: +1213333 ▼ (DID Range: +12133330000 - 0099)

PSTN Failover: None ▼

User Group: Executives ▼ [Go to this User Group](#)

Site: Headquarters ▼

Language: English ▼

Home Port:

IP Phones ShoreDoc IP Phone ▼
 Ports Bedrock - 8 ▼
 SoftSwitch Contact Center ▼

Current Port: ShoreDoc IP Phone Go Home

Jack #:

Mailbox on Server: ShoreDoc-HQ ▼ [Voice Mail Delivery and Notifications](#)

Accept Broadcast Messages

Include in System Dial By Name Directory

Make Number Private

Enable Fax Redirection

Enable Fax Server Integration

Allow Use of Soft Phone

Client Type: Operator ▼

Client User ID:

Client Password:

Voice Mail Password:

Email Address:

Conference Bridge:

Server: Collab ▼

User ID:

Password:

[Edit System Directory Record](#)

Figure 8-4 Edit User Page

8.5.2 Assigning IP Phones by User

You can also assign IP phones through the *Edit User* page, one user at a time. You can assign phones by specific IP addresses or assign users to Any IP Phone according to the user's needs.

To assign an IP phone:

Step 1 Click *Users > Individual Users* from the navigation frame.

Step 2 From *Add New User at Site*, select the site for the new user.

Step 3 Click *Go*. The *Edit User* page appears, as shown in Figure 8-4.

Step 4 Enter the information for the user you wish to add. For information about user settings, see Chapter 10, starting on page 173.

Step 5 From Home Port, click *IP Phones* and either select the IP phone by IP address or select *Any IP Phone*. Only the IP phones in the current site are in the list.

Step 6 Click *Save*.

8.6 Viewing and Editing IP Phones on the System

To help you manage the IP phones, ShoreWare Director allows you to view and edit all IP phones on the system. Figure 8-5 shows the *IP Phones* list and edit pages.

The screenshot displays the 'IP Phones' management page. At the top, there are filter options for 'By Sites' (set to 'All Sites') and 'By Switches' (set to 'All Switches'). Below the filters is a table with the following data:

Name	Site	Switch	MAC Address	IP Address	Current User	Home User	Phone Type
ShoreDoc IP Phone	Headquarters	Doc-24	00-04-F2-00-8E-B9	10.7.0.40	Adam Ant	Adam Ant	IP-100

An inset window titled 'IP Phone Info' provides detailed information for the selected phone:

- Name: ShoreDoc IP Phone
- Site: Headquarters
- Switch: Doc-24
- MAC Address: 00-04-F2-00-8E-B9
- IP Address: 10.7.0.40
- Current User: Adam Ant
- Home User: Adam Ant

Buttons for 'Save', 'Close', 'Previous', and 'Next' are located at the bottom of the inset window.

Figure 8-5 IP Phones List and Edit Pages

8.6.1 Viewing IP Phones

To view the IP phones on the system:

Step 1 Click *IP Phones > Individual IP Phones* from the navigation frame.

Step 2 From the *By Site* drop-down list, select the site that you want to view (or all sites).

Step 3 To view the IP phones connected to a specific switch, select the switch you want to view from the *By Switch* drop-down list.

The IP phones for the sites and switches you have selected are displayed.

IP phones can also be viewed from the maintenance pages of **ShoreWare Director**. For more information, see Chapter 18, starting on page 345.

8.6.2 Renaming IP Phones

You can change the name of an IP phone from the *IP Phone List*. By default, IP phones are listed by MAC address in the *IP Phone List Name* column.

To change the name of an IP phone:

Step 1 Click *IP Phones > Individual IP Phones* from the navigation frame.

Step 2 Select your viewing options so that the IP phone or phones you want to re-name are listed in the *IP Phone List*.

Step 3 From the *Name* column, click the IP phone you want to re-name. A dialog box displays the information for the IP phone you selected.

Step 4 Enter the new name in the *Name* text-entry field.

Step 5 Click *Save*.

8.6.3 Deleting and Moving IP Phones

IP phones can be deleted from the system or moved to another site.

To delete an IP phone:

Step 1 Click *IP Phones > Individual IP Phones* from the navigation frame.

Step 2 Select your viewing options so that the IP phone or phones you want to delete are listed in the *IP Phone List*.

Step 3 Click the check box of the IP phone you want to delete.

WARNING Make sure that you have selected the correct phone and that no other phones are selected.

Step 4 Click *Delete*. A dialog box requests confirmation.

Step 5 Click *Yes* to delete the phone. Click *Cancel* if you have selected the wrong phone.

If you wish to add the IP phone back into the system, you must reboot the IP phone. It will be reconfigured during the boot process and become available again. For more information on IP phone configuration, see the *ShoreTel 8 Planning and Installation Guide*.

To move an IP phone to a destination switch on a remote site, the remote site must have an IP address range defined. You may not move an IP phone to a switch on a remote site if the IP address of the phone is not within the IP address range of the destination site.

You may move an IP phone across switches at Headquarters without entering an IP address range for the Headquarters site. The IP address range restrictions are only for switches at remote sites.

To move an IP phone:

Step 1 Click *IP Phones > Individual IP Phones* from the navigation frame.

Step 2 Select the switch you are moving the IP phone(s) to from the drop-down list.

Step 3 Select your viewing options so that the IP phone or phones you want to move are listed in the *IP Phone List*.

Step 4 Click the check box of the IP phone you want to move.

Step 5 Click *Move*. A dialog box requests confirmation.

Step 6 Click *Yes* to move the phone. Click *Cancel* if you have selected the wrong phone(s).

8.6.4 IP Phone State Display

The following list explains the states displayed by the IP phones:

- **Available:** Phone has no user assigned to it. Calls can be placed from the phone but it does not receive calls. The Caller ID is “Anonymous”.
- **<User Name> <User Ext>:** The phone is assigned to <User Name>.
- **Anonymous:** The assigned user has used the Office Anywhere feature. Calls can be placed from the phone but it does not receive calls. The Caller ID is “Anonymous.”
Another method by which it can become anonymous is if the administrator explicitly configures anonymous phones which do not have assigned users.
- **Unavailable:** The phone was once in the ShoreTel 6.1 system but has been removed using Director. The phone has no dial tone and is not functional.

8.6.5 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 Press the *MUTE* key followed by 4636# (INFO#). The phone will display the first two parameters.

Step 2 Press # to advance the display or * to exit. The phone will resume normal operation after the last parameter has been displayed.

8.6.6 Resetting the ShorePhone

You can reset the phone by entering a key sequence from the phone’s keypad.

To reset the phone, press the MUTE key followed by 73738# (RESET#). The phone will reboot.

8.7 Loading Custom Ring Tones

ShoreTel IP phones offer four different sets of ring tones, with each set consisting of one tone for internal calls and another for external calls. The user is free to select from the four tones as desired. However, in some densely populated work environments, four tones may not be enough for users to be able to distinguish the sound of their phone from that of their neighbors' phones.

To help reduce confusion, most ShoreTel IP phones support the ability to load custom ring tones on an IP phone so that each user can have a unique ring tone.

One set consisting of two custom ring tones can be loaded onto each IP phone. This new set of tones displaces one of the existing sets of ShoreTel ring tones, as shown below.

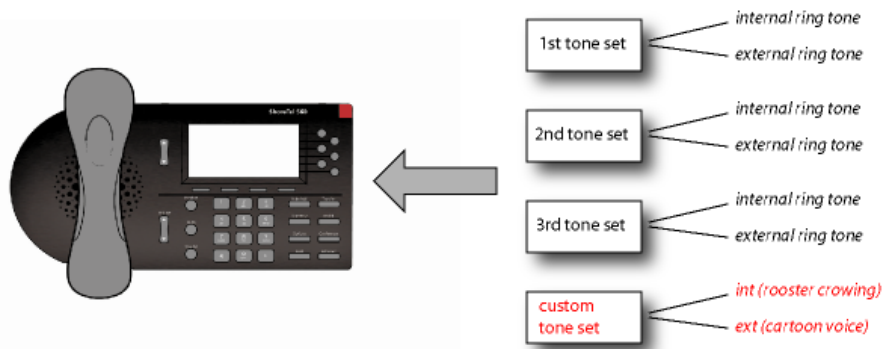


Figure 8-6 The 4th set of ShoreTel ring tones is replaced by the new custom tones

Ring tones must be in Microsoft Wave File format (i.e. .wav format). These custom ring tones are not provided by ShoreTel, but numerous web sites offer files that can be downloaded for free. A quick search should yield a large number of possibilities. Once a set of custom ring tones has been identified, the system administrator must load these ring tones onto a user's IP phone via an FTP server.

Details:

- This feature is supported on all ShoreTel IP phones, but is not supported on the Polycom IP100 phones.
- ShoreTel offers no tools for creating or managing the custom Wave Files.
- Custom files must be loaded in .wav format. Other file formats, such as MP3, are not supported.
- The following formats are supported by the phone:
 - μ -law: 8-bit, 8 kHz, 16 kHz, Mono
 - α -law: 8-bit, 8 kHz, 16 kHz, Mono
 - 16-bit, 8 kHz, Mono -or- 16-bit, 16 kHz, Mono
- Only two custom tones can be loaded on a single phone. The sum total file size of both tones must be less than 750KB. Files exceeding this size will not be downloaded.
- If the wave file has a length less than 6 seconds, the phone will loop the ring out to a 6-second length before it is repeated.

The process of loading ring tones onto an IP phone consists of the following steps:

Step 1 Identify Wave Files for use (via online search or by creating them manually). These files must be stored on a server that is accessible to the IP phone via anonymous FTP. (The server does not necessarily have to be the same as the one that is hosting the configuration files.)

Step 2 Manually create a phone-specific configuration text file and store it in the same directory as the standard IP phone configuration files. The name of the phone-specific file should contain the MAC address of the target phone where the ring tone will be loaded. The MAC address can be found on the sticker on the back of the phone. (See Section 8.7.1 below for details.)

Alternatively, you can load the same pair of custom ring tones onto several IP phones at the same time, but the phones must all be of the same type (i.e. all IP560g models or all IP212k models). This approach could cause ring tone confusion if the phones were concentrated in one area of the building.

Step 3 Reboot the phone so that it will pick up the information in the text file, along with the pointer to the Wave File. At boot time, the IP phone will display the success or failure of downloading the phone-specific configuration file and any configured Wave Files.

8.7.1 Creating the Phone-Specific Configuration File

To create a phone-specific configuration file, two new configuration parameters, *WaveRinger1* and *WaveRinger2* can be inserted into a phone-specific configuration file. These parameters identify the name and location of the custom ring tones that the IP phone will download (via FTP) into the phone's RAM at boot time.

WaveRinger1	Up to 64 ASCII Characters	Used to assign one Wave File to any of the ring tones defined in Table 2. The first value is the ring tone, and the second value is the location of the file on the FTP server.
WaveRinger2		
<p><i>Examples:</i></p> <p><i>WaveRinger1 L/rg 192.168.0.20/audio/dave.wav</i></p> <p><i>WaveRinger2 L/r1 192.168.0.20/audio/dave.wav</i></p>		

Table 8-1 WaveRinger1 and WaveRinger2 configuration parameters

The name of the phone-specific configuration file should be as follows:

shore_aabbccddeeff.txt

where "aabbccddeeff" is the MAC address.

To load the same custom ring tone onto several IP phones at the same time, do not use the MAC address in the name of the file. Instead, use the custom configuration file for the phone type being deployed (e.g. S6 for an IP560). For example, the configuration file name for an IP560 phone would be: S6custom.txt

For example, to load one of the custom ring tones, you could replace **L/r14** (i.e. Ring 4 External) and **L/r15** (i.e. Ring 4 Internal) with the name and location of the file containing the new custom ring tone.

Also note that it is possible to replace internal and external ring tones in separate sets (e.g. *Ring 2 external* and *Ring 4 internal*), but only one set of ring tones can be active at a time so activating either set of ring tones will only activate only one of the custom rings tones at a time.

Ring tone	Symbol
Standard - External ring	L/rg
Standard - Internal ring	L/r1
Ring 2 - External ring	L/r10
Ring 2 - Internal ring	L/r11
Ring 3 - External ring	L/r12
Ring 3 - Internal ring	L/r13
Ring 4 - External ring	L/r14
Ring 4 - Internal ring	L/r15

Table 8-2 Ring tones and symbols

8.8 Customizing Color LCD Displays with Wallpaper

The ShoreTel 265 and 565g IP phone models offer color TFT-LCD color displays. One of the benefits of having a color display is the ability to download images of your choice onto the phone from a server and display the image as wallpaper.

8.8.1 Wallpaper File Specifications

The wallpaper is 320 by 240 pixels and uses an uncompressed 256-color .bmp file format. Each of the 256 colors is defined by a 24-bit RGB value. Bitmap files can be composed using MS Paint or any other editor that can create Paint-compatible files.

To verify that a graphic file can be used as a wallpaper image:

Step 1 Open the image in MS Paint.

Step 2 Open the *Attributes* panel by selecting *Image -> Attributes* from the main menu.

Step 3 Verify the following parameters are set as follows and make the appropriate adjustments:

- Width = 320
- Height = 240
- Units = Pixels
- Colors = Colors

Step 4 Close the *Attributes* panel by clicking OK.

Step 5 Open the *Save As* panel by selecting *File -> Save As* from the main menu.

Step 6 Verify that *Save as* type is set to *24-bit Bitmap (*.bmp;*.dib)*.

Step 7 Note the proposed file name and directory, and click *Save*.

8.8.2 Downloading the File to Several Color-Screen IP Phones

The wallpaper file that each IP phone displays is specified in configuration files located in the phone configuration directory on the Headquarters server. On standard ShoreTel installations, the phone configuration directory is C:\netpub\ftpboot.

ShoreWare specifies one text file for each phone model that defines default characteristics for all phones of that model type on the system. You specify the default Wallpaper for phones of a specific model by adding a line to the corresponding configuration file.

To load a wallpaper image onto all system phones of an specific model:

Step 1 Save the wallpaper file on the headquarters (HQ) server in following directory: `C:/Inetpub/ftproot`

Step 2 Access the `C:/Inetpub/ftproot` directory on the HQ server.

Step 3 Open the customization file for the desired phone model:

- For IP 265, open `s36custom.txt`
- For IP 565g, open `s6ccustom.txt`

Step 4 Add the following line to the open file: `Wallpaper2pixmap abc.bmp`, entering the name of the wallpaper file in place of `abc.bmp`, then save and close the file.

For example, if the wallpaper file is name `logo.bmp`, then enter `Wallpaper2pixmap logo.bmp`.

Step 5 Open the configuration file for the desired phone model:

- IP 265: open `shore_s36.txt`
- IP 565g: open `shore_s6c.txt`

Step 6 Verify that the file contains the one of the following lines

- IP 265: `Include s36custom.txt.`
- IP 565g: `Include s6ccustom.txt.`

Add this line to the file if it is not present.

Step 7 Reset the phones.

Step 8 Verify that, when each phone reboots, they download and save the wallpaper file.

Step 9 Verify that each phone displays the new wallpaper file.

8.8.3 Downloading the File to a Single Color-Screen IP Phone

Individual configuration files override model default settings for individual IP phones. You can assign custom wallpaper files for individual IP phones by modifying the corresponding configuration file.

To load a wallpaper image onto a single phone:

Step 1 Save the wallpaper file on the HQ Server computer in following directory: `C:/Inetpub/ftproot`

Step 2 Access the `C:/Inetpub/ftproot` directory on the HQ Server.

Step 3 Create a text file named `shore_XXXXXX.txt`, where `XXXXXX` is the MAC address of the phone. Use lower case text when naming the file.

The MAC address is a twelve digit number that uniquely identifies each individual device. This address is printed on the white bar code located on the bottom of the phone.

Example: If the MAC address of an IP 565g is 00104907020C, then create a file named shore_00104907020c.txt

Step 4 Add the following line to the open file: `Wallpaper2pixmap abc.bmp`, entering the name of the wallpaper file in place of `abc.bmp`, then save and close the file.

For example, if the wallpaper file is name `logo.bmp`, then enter `Wallpaper2pixmap logo.bmp`.

Step 5 Reset the phone.

Step 6 Verify, while the phone reboots, that it downloads and saves the wallpaper file.

Step 7 Verify that the phone displays the wallpaper file.

8.9 VPN Phone

8.9.1 Introduction

8.9.1.1 Definition

VPN Phone that provides secure audio communications for ShorePhones located remotely from ShoreGear switches through open VPN SSL tunnels. To support VPN phone, ShoreTel provides two VPN Concentrators capable of supporting up to 100 calls through VPN tunnels. Three ShorePhones support the VPN Phone feature: IP565g, IP560g, and IP230g.

8.9.1.2 Terms

Secure Socket Layer (SSL): a cryptographic protocol that provide secure communications on the Internet.

Point to Point Protocol (PPP): a data link protocol used to establish a direct connection between two nodes over serial cable, phone line, trunk line, cellular telephone, specialized radio links, or fiber optic links

Virtual Private Network (VPN): a computer network in which some internode links are facilitated via open connections or virtual circuits through a larger network instead of via physical wires. The link-layer protocols of the virtual network are said to be tunneled through the larger network. One common application is secure communications through the public Internet.

VPN Concentrator: a gateway that provides secure access to a corporate network for remote devices through VPN tunnels.

Stunnel: a multi-platform program that provide SSL tunnels between a remote device and a VPN gateway.

Tunnel: an link between two networks (such as LANs) that are connected by a larger network (such as a WAN or the Internet). Tunnel packets containing messages exchanged by the smaller networks are encapsulated into packets that facilitate routing through the larger network. Each VPN communication session establishes a tunnel, which is removed when the session is finished.

8.9.2 Functional Description

8.9.2.1 Feature Summary

VPN Phone provides secure audio communications for ShorePhones located remotely from ShoreGear switches through open VPN SSL tunnels.

The feature includes an Open SSL VPN client in the ShoreTel IP phone and an Open SSL VPN Gateway. The Open SSL structure allows the traversal of firewalls implemented by many enterprises for blocking VPN tunnels.

8.9.2.2 Components

VPN Phone implementation requires two components – a VPN Concentrator and a ShorePhone capable of communicating over a VPN

VPN Concentrator

The SSL based VPN Concentrator enables remote ShorePhones to establish secure voice communications with through the local ShoreTel PBX through SSL VPN tunnels. For every tunnel, a virtual PPP interface is created on VPN Concentrator and a peer PPP interface is created on the remote ShorePhone. Signaling and media streams go through the PPP interface and are secured by SSL encryption.

ShoreTel offers two VPN Concentrator Models:

- VPN Concentrator 5300 – supports a capacity of 100 calls.
- VPN Concentrator 4500 – supports a capacity of 10 calls.

The VPN Concentrator 4500/5300 Installation and Configuration Guide provides additional information about the ShoreTel VPN Concentrators.

VPN Phone Licenses

ShoreTel licenses VPN Phone usage on a stunnel basis. Establishing a stunnel uses a license, which becomes available when the tunnel is discontinued.

ShorePhone

Three ShorePhone models support the VPN Phone feature:

- IP565g
- IP560g
- IP230g

8.9.2.3 Network Configuration

Figure 8-7 depicts the functional structure the ShoreTel implementation of a VPN network.

The VPN Concentrator is located at ShoreTel's Main Site, connected to the same LAN as local ShoreGear switches and the Main ShoreTel server. Refer to the VPN Concentrator 4500/5300 Installation and Configuration Guide for specific deployment options based on the router and firewall configuration of the ShoreTel Network.

8.9.2.4 Stunnel Establishment

Each remote device is assigned a username and password that is recognized by the VPN Concentrator. Phone logging into the Concentrator are authenticated through the verification of its username and password. When the phone is successfully authenticated,

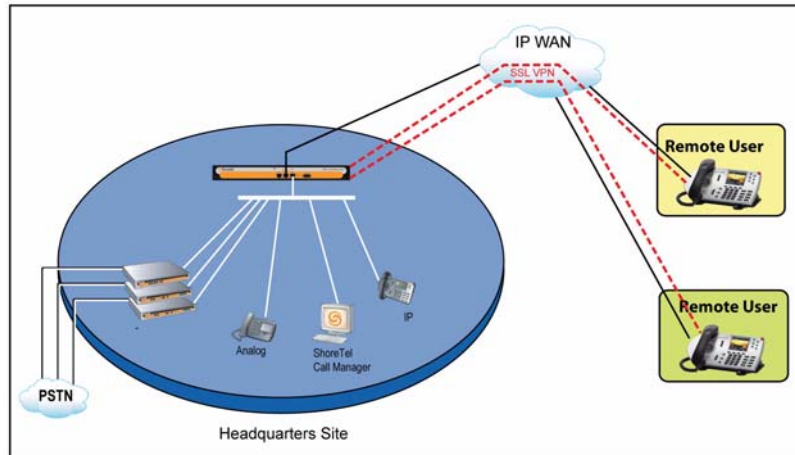


Figure 8-7 ShoreTel VPN Phone Implementation Architecture

the Concentrator establishes a stunnel to that phone, after which it can receive and make phone calls through the ShoreTel system. The stunnel remains in place until the phone sets the VPN parameter to off or the Concentrator times out all stunnel connections.

Establishing a stunnel requires an available VPN phone license. If the number of active stunnels equals the number of available licenses, The VPN Concentrator will not establish new stunnels until an existing stunnel is disconnected.

8.9.2.5 Phone Calls

After a stunnel is established from the ShorePhone to the Concentrator, VPN phone calls are performed from the ShorePhone in the same manner as if the phone is located on same LAN as the VPN Concentrator. The VPN Concentrator manages the connection from the phone to the ShoreTel.

8.9.3 Implementation

Configuring VPN Phone comprises three steps:

- Installing the VPN Concentrator
- Configuring Director
- Configuring the ShorePhone

The following sections describe the VPN Phone implementation process.

8.9.3.1 VPN Concentrator

Refer to the VPN Concentrator 4500/5300 Installation and Configuration Guide for instructions on physically inserting the VPN Concentrator into the network. The guide also describes web browser panels that configure the VPN concentrator. The following sections describe the panels and fields that require configuration.

Initial Configuration

The VPN Concentrator is shipped with the pre-configured IP address 192.168.1.1 for the LAN port.

To access the VPN Concentrator web interface panels, perform the following:

Step 1 Assign static IP address 192.168.1.2 with subnet 255.255.255.0 to the Ethernet interface of the computer that is connected to the LAN port.

Step 2 Launch a web browser on the PC and access the following URL

http://192.168.1.1.

The computer displays the login panel shown in Figure 8-8.

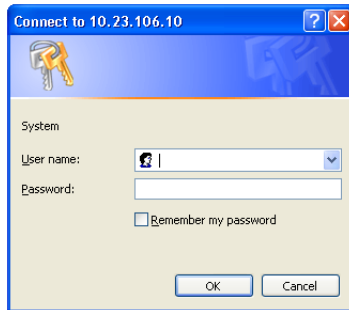


Figure 8-8 VPN Concentrator Login Panel

Step 3 Enter the following parameter values to log into the system

username – root
password – default

Network panel

The Network panel defines the IP address through which the VPN Concentrator is accessed. To access the Network panel, select Network in the blue Configuration Menu on the left side of the panel. Figure 8-9 displays the Network panel.

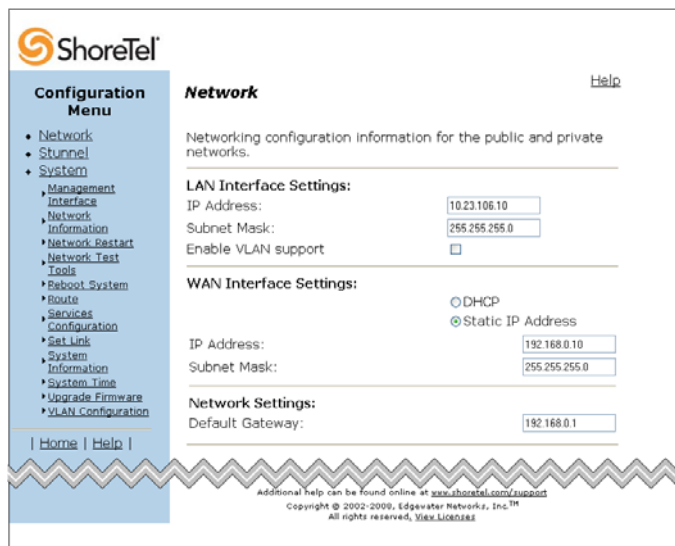


Figure 8-9 Network Configuration panel

Enter the appropriate values in the following data entry fields:

- LAN Interface Settings: Enter the IP address by which other LAN devices will access the VPN Concentrator

- WAN Interface Settings: Enter the IP address by which remote devices can access the VPN Concentrator.

Stunnel panel

The Stunnel panel specifies setup characteristics about the Stunnels that facilitate communications between the concentrator and the remote phones. To access the Stunnel panel, select Stunnel in the blue Configuration Menu on the left side of the panel. Figure 8-10 displays the Network panel.

Figure 8-10 Stunnel Configuration Panel

Verify that the following parameters are set as follows:

- Stunnel Enable is selected
- Stunnel Server IP Address is set to the LAN address of the VPN Concentrator, as specified on the Network panel
- Stunnel Server Port Number is set to 443.

Route panel

The Route panel defines the static routes to networks or servers on the LAN. To access the Route panel, select *System -> Route* in the blue Configuration Menu on the left side of the panel.

To add a static route, perform the following:

Step 1 Enter the subnet address and mask in the IP Network and Netmask data fields, respectively.

Step 2 Enter the IP address that accesses the Gateway server of the added network.

Step 3 Press the Submit button.

Figure 8-11 Route Configuration Panel

Usersnames' List panel

The Username Configuration Panel defines the user accounts that are allowed Stunnel access to the VPN Concentrator. To access the Route panel, select *Stunnel -> Username Database* in the blue Configuration Menu on the left side of the panel.

Figure 8-12 Username Configuration Panel

To add a user account, perform the following:

Step 1 Enter the username and password for the user in the Username and Password data fields, respectively.

Step 2 Re-enter the password in the Confirm Password data entry field.

The Password and Confirm Password data entry field contents must be identical.

Step 3 Press the Submit button.

8.9.3.2 ShoreWare Director

ShoreWare Director assigns codecs on the basis of the site assignment of the ShorePhone's IP address. Assigning the IP address block allocated to the VPN Concentrator to a specific site assures that the switch uses the proper codec when handling VPN Calls.

To assign an IP address block to a site, perform the following:

Step 1 Open the IP Phone Address panel in Director by selecting *Administration -> IP Phones -> IP Phone Address Map* from the Main Menu.

Step 2 Open the IP Phone Address Map Info popup, as shown in Figure 8-13, by pressing the New button in the top left corner of the panel.

The screenshot shows a web browser window titled "IP Phone Address Map Info - Microsoft Internet Explorer". The form contains the following fields and controls:

- Site:** A dropdown menu with "Remote" selected.
- Low IP Address:** A text input field containing "10.23.43.103".
- High IP Address:** A text input field containing "10.23.43.103".
- Caller's Emergency Service Identification (CESID):** A text input field containing "e.g. +1 (408) 331-3300".
- Teleworkers:** A checkbox that is currently unchecked.
- Buttons:** "Save", "Close", "Previous", and "Next".
- Footer:** "© 1998-2007 ShoreTel, Inc. All rights reserved."

Figure 8-13 IP Phone Address Map Info panel

Step 3 Enter the Site Name and the low and high IP address of the block allocated to VPN calls in the appropriate data entry fields.

The Site Name parameter specifies the codecs used for call involving devices at the specified IP addresses.

Step 4 Press the Save button.

Refer to Table B-1 on page 406 for configuration recommendations for Emergency 911 related features on VPN phones

8.9.3.3 Phones

ShorePhones are manually configured to establish a stunnel with the VPN concentrator. After the phone is configured and placed on a WAN port (such as the internet) it attempts to communicate with the Concentrator. ShoreWare does not provide DHCP options for automatically setting these parameter values at startup.

The following parameters are manually configured on the ShorePhone to perform VPN calls:

- **VPN Gateway:** This parameter specifies the WAN IP Address of the VPN Concentrator to which the ShorePhone connects. Default value is *0.0.0.0*.
- **VPN Port:** This parameter specifies the port number of the VPN Concentrator to which the ShorePhone connects. Default value is *443*.

- **VPN:** This parameter, when set to *On*, enables VPN Phone on the ShorePhone. Default setting is *Off*.
- **VPN User Prompt:** This parameter, when set to *On*, programs the ShorePhone to prompt the user for a VPN user name after completing a power cycle.
- **VPN Password Prompt:** This parameter, when set to *On*, programs the ShorePhone to prompt the user for a VPN password after completing a power cycle.
- **FTP:** This parameter specifies the IP Address of the RTP server from which the phone requests VPN Phone software upgrades. When set to the default value of 0.0.0.0, the phone solicits upgrades from the IP Address of the VPN Gateway.

Manually Configuring a ShorePhone

To configure the ShorePhone IP phone, perform the following:

Step 1 Press the Mute button, then enter SETUP#

SETUP translates numerically to 73887

Step 2 Enter the phone's password, followed by #.

Step 3 Press the # button to step through the phone options.

Configuring the User Name and Password

The user name and password is stored in non-volatile RAM on the phone. Power cycling and normal phone operations have no effect on the stored name and password. The VPN Concentrator authenticates the ShorePhone when the phone attempts to establish a stunnel by verifying the phone's username and password is included in the user accounts on the Usernames' List panel.

New ShorePhones are shipped with this memory location vacant. The first time a user power cycles the phone with the VPN parameter set to *On*, the phone prompts the user for a username and password. The phone prompts for these values if the **VPN User Prompt** and **VPN Password Prompt** parameters are set to *On*; otherwise, the ShorePhone continues using the previous memory contents when attempting to establish a stunnel.

Setting Call Control Options

9.1 Overview

This chapter provides information about configuring the system-wide call control features of the ShoreTel 6.1 system. You can configure:

- Account Codes
- Bridged Call Appearances
- Hunt Groups
- Paging Groups
- Pickup Groups
- Route Points
- Ringdown Circuits
- Bandwidth Management
- Codec Profiles
- Media Encryption

Clicking the *Call Control* link in the navigation frame provides access to other *Call Control selections*, shown in Figure 9-1.

9.2 Account Codes

Account codes are typically used to assist ShoreTel users in the billing of their clients. For example, if a law firm wants to keep track of the length of calls to their clients so that they may later bill those client for services rendered, they can enter an account code that corresponds to that client before dialing the client's phone number. At the end of the call, the call length, time, and date are entered in a record, thus helping the firm to keep track of the calls made to each of their clients.

Account Codes can vary in length and be flexibly formatted. You can configure the system to require users to enter an account code for outbound calls in a mandatory or optional fashion. In this way, the account code can also function to prevent unauthorized employees from dialing long-distance numbers.

ShoreTel 6.1 supports wildcard characters in account codes. This enhancement allows the system to surpass the previous limit of 50,000 account codes so that an almost unlimited number of account codes can be supported. The wildcard character – a question mark – can be entered in place of DTMF digits in the account code.

The use of wildcards introduces less strict validation of the account code entered by the user. Rather than checking each individual code, a length check is performed instead. The introduction of wildcards into the account codes does not impact the ability of the system to assign an account code to individual clients.

ShoreTel
ShoreWare Director

Logoff Tee

Administration

- Users...
- Trunks...
- IP Phones...
- Switches
- Call Control...
 - Account Codes
 - Bridged Call Appearances
 - Hunt Groups
 - Paging Groups
 - Pickup Groups
 - Route Points
 - Options
- Voice Mail...
- Auto-Attendant Menus
- Workgroups
- Schedules
- Personal Call Manager...
- System Directory
- Application Servers
- Conference Bridges
- Sites
- System Parameters...
- Preferences

Maintenance

Reporting

Documentation

Call Control Options
Edit

Save Reset

Edit this record Refresh this page

General:

Use Distributed Routing Service for call routing.

Enable Monitor / Record Warning Tone.

Generate an event when a trunk is in-use for minutes.

Park Timeout (1-100000) after seconds.

Hang up Make Me Conference after minutes of silence.

Delay before sending DTMF to Fax Server: msec

Enable SIP Session Timer.

Session Interval (90 - 3600): sec

Refresher: ▾

Voice Encoding and Quality of Service:

Intra-Site Calls: ▾

Inter-Site Calls: ▾

FAX and Modem Calls: ▾

Maximum Inter-Site Jitter Buffer: msec

DiffServ / ToS Byte (0-255):

Admission control algorithm assumes RTP header compression is being used.

Enable Media Encryption.

Always Use Port 5004 for RTP.

Trunk-to-Trunk Transfer and Tandem Trunks:

Hang up after minutes of silence.

Hang up after minutes.

Figure 9-1 Call Control Options Edit panel

You can create account codes with non-numeric characters, but these characters are discarded during code collection. The following table gives example account codes and shows how the Account Codes Service interprets the code.

Sample Account Code	Recorded Code
Sales 200	200
1001-3	10013
1.234A	1234
3000 Exec 2	3002

Table 9-1 Sample Account Codes

Account code collection is enabled on the basis of user groups, with the collection of account codes set to be one of three states:

- Disabled
- Optional
- Forced

For information about account codes and User Groups, refer to Section 10.3 on page 185.

Call Detail Reports include details of account codes associated with outbound calling. Account Codes are associated with a configurable extension and have a dedicated user group that defines ultimate call permissions and trunk group access.

A new user group named “Account Codes Service” is created for use by the Account Codes Service. Since it is only intended for use by the Account Codes Service, this user group does not appear in drop-down lists for the assignment of User Groups to users and other objects

such as workgroups. You can, however, change all attributes of the Account Codes Service User Group except the fields indicating whether Account Codes are disabled, optional, or required. Refer to Section 7.2.1 on page 96.

When account code collection is enabled or required for a user group, calls placed via the telephone or the Call Manager are routed to the account code extension. The Account Codes Service prompts the user to enter an account code followed by the “#” key. If the account code entered does not match the digits in a stored account code, an explanation message is played and the user can enter an account code again. When a matching account code is collected, the call is placed according to the originally dialed number.

The call permissions define which dialed numbers will be directed to the Account Codes Service for user groups configured with account codes. For calls that are redirected to the account codes extension, the call will be completed with the trunk access and call permissions of the Account Codes Service.

This structure imposes two sets of permissions to outbound calls:

1. The call permissions for the user group of the user that places the call are used to determine whether an account code must be collected or not.
2. The call permissions for the Account Codes Service user group determines whether calls are finally placed or if the intercept tone is played.

Calls forwarded to external numbers are NOT subjected to account code restrictions. Forwarding calls to external numbers is controlled from the Class of Service settings. For more information, see Section 10.2 on page 174.

The Account Codes Service is associated with a system extension that is hosted on the SoftSwitch running on the headquarters (HQ) server only. If the HQ SoftSwitch is not reachable by the originating ShoreGear switch, the call will be handled according to the setting on the caller's user group. Specifically, during such a connectivity outage, calls placed by users who have optional account code collection will be automatically placed and calls placed by users who have forced account code collection will be automatically rejected.

If you have a ShoreTel Converged Conference solution installed, Account Codes can be set up for collaborative voice and data conferencing. Document sharing is a feature allowed in a 2-way or Make Me Conference call. Up to 20 documents can be dragged and dropped during the call. You can create associated project codes on the conference bridge for tracking purposes. Bridge project codes are not enforced.

9.2.1 Adding and Editing Account Codes

To configure the account code wildcard feature 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 **Call Control** link to display a window similar to the one shown below:

The top portion of the Account Codes page offers you two options for filtering the account codes list. When you have defined one or both of these filters, click **Apply**. To save your filter, click **Save Filter**. Clicking **Clear** clears both fields.

The filtering options are:

Account Codes [Help](#)

Filter Account Code Find Now Clear

By Name:

By Account Code:

Account Code List 0 records checked.

Delete New...

<input type="checkbox"/>	Name	Account Code
<input type="checkbox"/>	blah	1rob
<input type="checkbox"/>	Marketing	?????
<input type="checkbox"/>	Robert1	1234
<input type="checkbox"/>	robert2	2ab
<input type="checkbox"/>	Sales	????
<input type="checkbox"/>	Sales 2	13256

Figure 9-2 Account Codes Page

- **By Name:** You can filter the list by the name given to the account codes. Enter partial account names to return a list of all accounts beginning with matching characters.
- **By Account Code:** You can filter the list by account code. Enter partial account codes to return a list of all accounts beginning with matching digits.

The bottom portion of the Account Codes page lists the current account codes and allows you to create and delete account codes. To delete an account code, click the check box to the left of the account code and click Delete.

Step 4 Click on the New button. A window appears, as shown in Figure 9-3, that allows you to enter the name and code of a new account.

Figure 9-3 Account Code Info dialog box

Step 5 Enter a name for the account code in the Name field.

Step 6 Enter an account code in the Account Code field. Account codes can be any length to a maximum of 15 digits. You can use prefixes to group account codes. For example, the account codes 1234 and 12345 can coexist. If your system will contain more than 50,000 account codes, you can enter a string of wildcard characters (a question mark) in place of the DTMF digits.

Up to 15 wildcard characters can be entered in an account code, but if all of the account codes in your organization have 6 digits, then you should enter 6 wildcards in this field. A single account code cannot contain a mix of digits and wildcard characters.

Step 7 Click *Save* to create the account. To create another account, click *Next*.

Step 8 Click *Close* to exit the Account Code Info dialog box.

Account codes are enabled on the basis of user groups. For more information, see Section 10.3 on page 185.

9.3 Bridged Call Appearances

A Bridged Call Appearance (BCA) is an extension that is shared among multiple users. A BCA is characterized by an extension number and its call stack depth. The extension number defines the method of contacting BCA users. The call stack depth specifies the number of call that can simultaneously reside on the BCA. The maximum BCA stack size is 24. Bridged Call Appearances are configured by the system administrator.

Each BCA is assigned to a switch and do not require a user license. BCAs support call handling but are not controlled by schedules.

Users answer BCA calls by pressing a ShorePhone button that is assigned to an existing BCA call stack position. Calls to a BCA occupy distinct call appearances and are identified by their position in the call stack. ShorePhone buttons that answer calls are configured to handle calls to a specific call stack position of a BCA. A button can be programmed for each position in the call stack.

Example: The system administrator configures a BCA with an extension of 118 and a call stack depth of 3. The ShorePhones of three users are configured to handle calls to the BCA as follows:

- User One has one button that answers calls from Stack Position #1.
- User Two has one button that answers calls from Stack Position #2.
- User Three has three buttons configured to answer BCA calls – the first button answers calls to Stack Position #1, the second button answers calls to Stack Position #2, and the third button answers calls to Stack Position #3.

The first incoming call to the BCA arrives on Stack Position #1. User Two cannot answer this call. A second call to the BCA will arrive on Stack Position #2 if the first call is still active. User One cannot answer that call. User Three is the only user that can answer calls that arrive on Stack Position #3.

When a call stack position on a BCA receives a call, all buttons configured for that stack position indicated an incoming call. When the call is answered, the LED on the phone of the person that answered the call becomes solid green while the other BCA stack buttons are red.

Users place calls through a BCA by pressing a programmed ShorePhone button. The LED on the outbound caller's phone becomes solid green, and the buttons associated with the BCA stack position on all other phones become solid red. If the call is placed on hold, the buttons associated with LEDs on all phones indicate a call on hold.

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.

A caller ID number can be associated with a BCA. The following rules determine which caller ID number is displayed for outbound BCA calls:

- Outbound to an internal extension – the name and number of the user that initiated the BCA call is sent. If the user is a “private”, the caller-id is blank.

- Outbound to an external number – the system sends the first number in the following list that is available:
 - Outbound caller-id number assigned to the BCA
 - DID number assigned to the BCA
 - External identification or caller-id number of the user initiating the BCA call
- Outbound to an external emergency number (such as 911) – the emergency identification or the user's CESID number is sent.

The system can be configured to display caller ID on inbound calls, in addition to enabling, disabling, or delaying inbound call ringing.

9.3.1 Switch Support for Bridged Call Appearances

ShoreGear 1-U Half Width and 1-U Full Width voice switches support BCAs, with the following limits:

- Up to 24 BCA extensions can be configured per switch
- Up to 128 BCA extensions (on other switches) can be monitored.
- A maximum of 32 phones can be configured to point to the same BCA extension.

You cannot configure BCA extensions IPBX voice switches. These switches can monitor up to 64 BCA extensions that reside on newer switches.

9.3.2 Configuring BCA Parameters

You configure BCAs from the Edit Call Appearance window, as shown in Figure 9-4. The following is a description of BCA parameters:

Figure 9-4 Configuring details associated with new BCA

- **Name:** This field specifies the label by which other Director panels and ShoreWare devices identify the BCA.
- **Extension:** This field specifies the extension through which the BCA receives inbound call.

- **Backup Extension:** This field specifies the extension that receives calls to the BCA when the switch that supports the BCA is out of service.
When using another BCA as the Backup Extension, the two BCAs and the must be assigned different switches.
- **DID:** This field specifies the DID number assigned to the BCA.
- **DNIS:** This field specifies the DNIS information sent on outbound BCA calls.
- **Outbound Caller ID:** This field specifies the Caller ID number that the system sends on outbound BCA calls.
- **Include in System Dial By Name Directory:** When this parameter is enabled, the BCA extension is available through ShorePhone display by pressing the Directory button.
- **Make Number Private:** This parameter designates the BCA extension as Private, as described in Section 10.4.2 on page 188.
- **Switch:** This field specifies the ShoreGear switch to which the BCA is assigned.
- **Call Stack Depth:** This field specifies the number of calls that the BCA can simultaneously handle.
- **No Answer Number of Rings:** This field specifies the No Answer condition for a call to a BCA.
- **Call Forward Destination:** These fields specify the recipient of unanswered inbound calls.
 - *Call Stack Full:* This number receives calls that are unanswered when the BCA is handling the number of call specified by the Call Stack Depth setting.
 - *No Answer:* This number receives calls that are unanswered after the number of rings specified by the No Answer Number of Rings setting.

To create a new, or edit an existing Bridged Call Appearance:

Step 1 Open the *Bridge Call Appearance List* window in ShoreWare Director by selecting *Administration -> Call Control -> Bridged Call Appearances* from the Main Menu.

Figure 9-5 displays the *Bridged Call Appearances List* window:

<input type="checkbox"/>	Name	Extension	Switch
<input type="checkbox"/>	New Bridged Call Appearance	4130	24-SG
<input type="checkbox"/>	New Multiple Extension Appearance	4500	24-SG
<input type="checkbox"/>	test MEA	4499	Fuji24

Figure 9-5 Setting up a new BCA

Step 2 Open the *Edit Bridged Call Appearances* window by either clicking the name of an existing BCA or click the New button.

Figure 9-4 displays the *Edit Bridged Call Appearances* window.

Step 3 Enter the required information

Step 4 Click **Save** to store your changes.

9.3.3 Configuring Answer Options

Users answer BCA calls by pressing IP Phone Buttons, which are configured to answer calls to a specific stack position of an BCA extension. IP Phone Buttons can also be configured such that a BCA call is answered when the user lifts the handset or presses either the speaker or headset buttons. The Program IP Phone Buttons panel, shown in Figure 9-6, programs an IP Phone button to answer a BCA call.

Program IP Phone Buttons
Headquarters User3

Save Reset

Edit Custom Keys Refresh this page

IP Phone Type: IP212k/230/265/530/560/560g/565

#	Function	Label	Target
1	All	Call Appearance	
2	Telephony	Bridged Call Appearance	BCA1

Extension: 118:BCATest1 Search

Call Stack Position: 1

Ring Delay Before Alert: <None>

Show Caller ID on Monitored Extensions:
 Never Only When Ringing Always

Enable Auto-Answer When Alerting

No Connected Call Action:
 Answer Only
 Dial Tone
 Dial Extension: Search
 Dial External: (e.g. 9+1 (408) 331-3300) Search

Figure 9-6 Programming an IP Phone button to answer BCA Calls

BCA parameters that are configured when configuring an IP Phone button include:

- **Extension:** This parameter specifies the BCA for which the button answers calls.
- **Call Stack Position:** This parameter specifies the individual calls to the BCA extension that the IP Phone Button can access.
- **Ring Delay Before Alert:** This parameter specifies the period that an inbound call is present on the IP Phone button before the phone plays an audio alert. Valid settings include:
 - *None:* The phone plays an audio alert on the first ring.
 - *1, 2, 3, or 4:* The phone begins playing the audio alert after the specified number of rings.
 - *Don't Ring:* The phone never plays the audio alert.
- **Show Caller ID on Monitored Extensions:** These options control the information that the phone user can see about the inbound call.
- **Enable Auto-Answer when Alerting:** When this option is selected, the Inbound BCA Call is answered by pressing the IP Phone button, taking the handset off-hook, pressing the speaker button, pressing the headset button, hook-flashing, or by pressing a unused Call Appearance button.
- When this option is not selected, the inbound BCA call is answered by pressing the IP Phone button programmed for the BCA or by pressing the Answer button.
- **No Connected Call Action:** These radio buttons programs the phone's ringdown behavior. Refer to Section 9.10.3.1 on page 166 for more information about configuring Ringdown.
 - **Answer Only:** Select this option to disable ringdown

- **Dial Tone:** Select this option to configure the phone as the recipient on a ringdown circuit.
- **Dial Extension:** Select this option to configure the button as the initiating end of a ringdown circuit when the recipient is an IP phone located on the ShoreWare network.
- **Dial External:** Select this option to configure the button as the calling end of a ringdown circuit when the recipient is a device not located on the ShoreWare network.

To access the IP Phone Buttons panel, perform the following:

Step 1 Select *Administration -> Users -> Individual Users* from the Director menu.

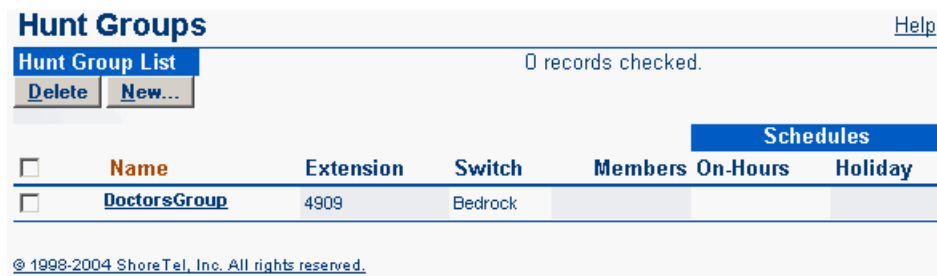
Step 2 Select the desired user on the **Individual Users** list.

Step 3 Select *Personal Options* on the panel bar at the top of the window.

Step 4 Select **Program IP Phone buttons**, located in the middle of the panel.

9.4 Hunt Groups

The *Hunt Groups* list page is shown in Figure 9-7.



Hunt Groups					
Hunt Group List					
0 records checked.					
Schedules					
<input type="checkbox"/>	Name	Extension	Switch	Members	Holiday
<input type="checkbox"/>	DoctorsGroup	4909	Bedrock		

Figure 9-7 Hunt Groups List panel

Clicking on a group name shows the details for the hunt group, as shown in Figure 9-8 in the Hunt Groups edit page.

Hunt groups allow a call to be offered to a limited set of user extensions with no reporting, queuing, sophisticated schedules, log-in, log-out, or wrap-up states. Each hunt group is composed of an ordered list of no more than 16 users. A maximum of 8 hunt groups totalling no more than 16 members can be assigned to a single switch. If your requirements more complex, you should use workgroups.

Rather than being reliant on the Headquarters Server, a hunt group can be assigned to the switch closest to the agents and/or trunks associated with it. The switch controls the hunting, with no dependency upon the server. Hunt Groups have an extension number and, optionally, can also have a DID and/or DNIS number. They can be call forward extensions for users, workgroups, route points, personal assistants, site fax redirect extensions, site operator extensions, and the target for trunk groups. They are also allowed as the backup destination for workgroups and route points. This can be useful to allow some basic call handling when the workgroup server is not reachable.

The caller ID displayed for a hunt call is the external caller's ID.

Hunt Groups

Edit Hunt Group

[Help](#)

New Copy Save Delete Reset

Edit this record
[Refresh this page](#)

Name:

Extension:

Backup Extension: Search

DID: +140896 (DID Range: +14089621300 - 21319)

DNIS: Edit DNIS Map

Include in System Dial By Name Directory

Make Number Private

Switch:

Call Stack Depth:

Distribution Pattern: Top Down Simultaneous

Rings per Member:

No Answer Number of Rings:

Call Member When Forwarding All Calls

Skip Member If Already On a Call

Call Forward Destinations:

Call Stack Full: Search
If no destination is specified, busy tone is played

No Answer: Search
If no destination is specified, hunting continues forever

On-Hours Schedule:

Holiday Schedule:

Off-Hours / Holiday Destination: Search

Current Call Handling Mode:

Filter Users By:

First Name: Last Name: Extension:

Sort By: Apply

Choose Members:

Show Page: < << >> >

2112 - ext1 ext1 2113 - ext2 ext2 2115 - user1 user1 2117 - ext4 ext4 2120 - Robert User2 2121 - Robert User3 2122 - Robert User1 2123 - Robert IP1 2128 - user3 user3 2129 - ext5 ext5	Add >> << Remove Move Up ^ Move Down v
--	---

Hunt Group Members:

2114 - ext3 ext3
 2116 - user2 user2

Figure 9-8 Configuring Hunt Groups

A user may belong to more than one hunt group. In addition, a user assigned to a workgroup may also be assigned to hunt groups. Each call is hunted as a new call; that is, if the hunt mode is top down, each new call begins hunting from the top of the list. In this case, the person at the top of the list will get most of the calls.

Fields are defined as follows:

- **Name:** This is the name of the hunt group. Each hunt group name must be unique. This is a required field.

- **Extension:** This is the extension number of the hunt group. Each hunt group extension must be unique.
This is a required field.
- **Backup Extension:** This is the backup extension of the hunt group. If the hunt group is unreachable or the switch is down, calls can be directed to this extension. A backup extension may be another hunt group, a workgroup, a route point, or a user.
This is a required field.
- **DID:** You can assign one DID number to a hunt group.
Check the first box to select DID. Make a selection from the drop-down list of area codes.
This is an optional field.
- **DNIS:** The *Edit DNIS Map* button invokes the *Select DNIS Trunk Group* dialog box. This lets you select a trunk group for DNIS routing. Only trunk groups that are configured for DNIS will be presented in the dialog box. You can assign multiple DNIS numbers to a hunt group.
DNIS is typically used to route 800-number calls to a workgroup or application.
This is an optional field.
- **Include in System Dial By Name Directory:** This check box includes the hunt group in the auto-attendant dial-by-name directory. No name is recorded for a hunt group. When a hunt group is chosen, the extension is announced by a generic message.
This is an optional field.
- **Make Number Private:** This check box makes the hunt group extension private. When the hunt group is private, it is not listed in the system directory and does not appear in the Call Manager dialing lists. See also Section 10.4 on page 187.
This is an optional field.
- **Switch:** Select from the drop-down list of available switches. This is the switch that will host the hunt group and do the hunting of calls
- **Call Stack Depth:** This lets you specify the maximum number of simultaneous calls that can be “stacked” on the hunt group extension. When this number is met, additional inbound calls will be routed to the Busy Destination.
Default value is 8. Valid entries are 1 through 16.
See also Section 10.2.1 on page 175.
- **Distribution Pattern:** Click either Top Down or Simultaneous. Top Down hunts sequentially through the ordered list of group members. Simultaneous rings all group members at the same time. The first to answer is presented with the call. The default is Top Down.
This is a required field.
- **Rings Per Member:** The default is 2 rings. All group member extensions ring with the same number of rings. If the phone is not answered, the hunt continues on to the next group member.
This is a required field.

- **No Answer Number of Rings:** The default is 6 rings. This value determines the number of ring backs a caller will hear while the call is being hunted. Once this value is exceeded, the call is sent to the No Answer Destination.

This is a required field.

- **Call Member When Forwarding All Calls:** Default is disabled. When enabled, even if a group member's call handling mode is Call Forward Always, the call is offered to the member.
- **Skip Member if Already on a Call:** Default is disabled. When enabled, even if a group member's call stack is not full, if the member's phone is busy or currently being offered a call, the new call is not offered to the member.
- **Busy Destination:** An alternate call destination can be specified for calls to be sent when all members of the hunt group are busy and the call stacks are full.

This is an optional field.

- **No Answer Destination:** An alternate call destination should be specified for times when no member answers a call. The hunt will continue until the No Answer Number of Rings value is exceeded, after which callers are sent to this No Answer Destination.
- **On-Hours Schedule:** From the drop-down list, select an on-hours schedule or None. Selecting None causes all calls to be treated as if it is on-hours.
- **Holiday Schedule:** From the drop-down list, select a holiday schedule or None. Holidays are handled the same as off-hours.
- **Off-Hours/Holiday Destination:** Each hunt group can have a call forward destination for use during the off-hours state.

This is an optional field.

- **Current Call Handling Mode:** You can configure several call handling modes for the hunt group: *On-Hours*, *Off-Hours*, or *Holiday*. The default is *On-Hours*. Consult Chapter 14, starting on page 293. This is an informational field.
- **Choose Members:** Click a member name and then click *Add* to add a member to the Hunt Group. Members can be removed and re-ordered, as well. This is useful since the Hunt Group membership is an ordered list.

Note that you can use filters to sort available members into an order that makes your selection process easiest for you.

9.4.1 Setting the Hunt Group to Busy

Users with correct permissions can set the hunt group to a busy state from the Switch Maintenance page or from the telephone user interface by using a star code (*18) followed by the hunt group extension. A confirmation prompt is played to confirm the state of the hunt group following entry of the star code. The *18 code is used to place the hunt group back into service, as well.

You can busy out the hunt group when all members are unavailable. Calls are then forwarded to the Busy Destination. After a switch reboots, the hunt group is available, by default.

Hunt Groups may also be busied out or returned to service from the Switch Maintenance page. From the Maintenance Quick Look page, select the switch that handles the hunt group, select a hunt group, and then change the busy state.

9.5 Paging Groups

As an alternative to using an in-house paging system, you can broadcast a message over a group of speakerphones with the Paging Groups feature. This feature allows a system administrator to designate groups of extensions that can be paged by dialing a single system extension and recording your message. This feature can be a cost-effective alternative for environments that do not already have an overhead paging system installed.

For environments that have an overhead paging system, Paging Groups can target your message to a select group of individuals within the organization while not exposing the message to everyone in the building, as would happen with an overhead page.

9.5.1 Adding Overhead Paging to Paging Groups

A Paging Extension (i.e. the extension which, when dialed, sends a page announcement to a site's overhead paging system) can be included in a Paging Group. By merging a Paging Extension within a Paging Group, you can broadcast a message to a select group of user extensions AND send it to the overhead paging system at the same time. Adding multiple Paging Extensions to an extension list provides the capability to simultaneously page the overhead paging system of multiple sites.

- Paging Group messages sent to an IP phone that is on-hook are announced on the speaker of that IP phone. Pages sent to an IP phone or analog phone that is already on a call are treated as a normal call and are routed to voicemail.
- Call handling does not apply to paging calls.
- A maximum of 100 extensions can be paged at one time.
- After receiving a request to play a paging message, the workgroup server introduces a short pause to synchronize the audio across several phones. This pause (which can last up to 6 seconds) allows calls to all affected extensions to connect to the server, after which the server begins playing the message.

This pause synchronizes the audio paging across a group of devices, reducing the chance of a cluster of phones playing the message at different times (which could create a disconcerting echoing effect if the phones were within earshot of one another). Refer to Product Bulletin 0200 on the ShoreCare website for more information.

9.5.2 Configuration

To specify who should be paged, create an extension list of paging user numbers. For information about creating extension lists, refer to Section 10.7 on page 230. Users in the extension list used as a paging group must belong to a user group that allows overhead paging. As an alternative to calling the paging number, group paging is available from the auto-attendant (for internal users) and can be used if this fulfills your paging needs. Group paging is not available to external callers.

Paging groups are viewed, edited, or deleted on the Call Control *Paging Groups* list page.

<input type="checkbox"/>	Name	Extension
<input type="checkbox"/>	AlternatePaging	4910
<input type="checkbox"/>	CentralPaging	4911

Figure 9-9 Paging Groups List Page

Select an existing group for editing or click New to add a new paging group. The *Paging Groups* edit page is shown in Figure 9-10.

Figure 9-10 Edit Paging Group panel

The fields on the Paging Groups edit page are:

- **Name:** This is the name of the paging group. Each paging group name must be unique. This is a required field.
- **Extension:** This is the extension number of the paging group. Each paging group extension must be unique. This is a required field.
- **Include in System Dial By Name Directory:** This check box includes the paging group in the auto-attendant dial-by-name directory. No name is recorded for a paging group. When a paging group is chosen, the extension is announced by a generic message. This is an optional field.
- **Make Number Private:** This check box makes the paging group extension private. When the paging group is private, it is not listed in the system directory and does not appear in the Call Manager dialing lists.
See also Section 10.4 on page 187. This is an optional field.
- **No Answer Number of Rings:** The default is 2 rings. This is always used for analog phones. It is used for IP phones if the phone is busy. This is a required field.
- **Extension List:** Select the name of the extension list to be used as the paging group from the drop-down list.

9.6 Pickup Groups

Pickup Groups is a traditional PBX and key system feature used in group environments that allows users to answer any ringing phone in that group. The feature works best in places where a set of people work together on a daily basis, such as design firms. If a group member is away from her desk and across the room when her phone rings, she can quickly answer the call from another person's IP phone by pressing the relevant soft key or programmable button, or by using a simple star command (*13 + extension) from an analog phone.

Similarly, if she is out of the office and her phone rings, anyone can answer her call from another phone with a simple 'group pickup' command and take a note for her.

User extensions are added to an extension list and then this list is associated with a pickup group. The pickup group has its own extension (e.g. x3755), but this extension is invalid, cannot be dialed, and thus acts more like a code than an actual dialable extension.

- Pickup groups can be associated with a programmable toolbar button, or with a programmable button on an IP phone, or on IP phones that have soft keys.
- The user whose phone will be picked up must have class of service “Call Pickup Allowed” to use this feature. However, other users need not be members of the pickup group to pickup a call.
- This feature is not supported on the following legacy ShoreTel switch models: ShoreGear T1 and ShoreGear E1.
- The pickup feature will support:
 - 24 members per group
 - 16 groups per switch
 - The members assigned to all pickup groups on a switch cannot exceed 80
 - A single user can be a member of up to 5 pickup groups
- A single switch can host a combined total of up to 24 hunt groups, bridged call appearances, and pickup groups.
- This feature can be accessed in three different ways:
 - **IP Phone** – If a programmable button has been configured for this feature, the user can press the button, or key, and enter the extension for the pickup group to answer the call.
 - **PCM** – If one of the pre-programmed buttons in PCM has been set up for pickup groups, a user can enter the extension of the group to answer the call. If the key has already been programmed with the extension of the pickup group, then it is not necessary to enter the extension.
 - **Analog Phone** – The user can enter the *13 command from the keypad, followed by the pickup group extension to answer calls from an analog phone.

9.6.1 Configuring Pickup Groups

Configuring the Pickup Groups feature consists of two separate tasks. First, you must create an extension list and populate it with the extensions of the members that will belong in this group. Second, you must create and name the pickup group and associate it with the extension list you just created.

To create the new extension list for a pickup group:

- Step 1** Launch **ShoreWare Director** and enter the user ID and password. Then click the **Login** button.
- Step 2** Click on the **Administration** link to expand the list (if it has not already been expanded).
- Step 3** Click on the **Users** link and then click on the **Extension Lists** link to display the **Extension Lists** window.
- Step 4** Click the **New** button to display a window similar to the one shown in Figure 9-11:
- Step 5** Enter the desired name in the **Name** field.
- Step 6** Select one or more users from the list on the left pane, and click the **Add** button to move them over to the **Extension List Members** pane on the right.

Figure 9-11 Creating a new extension list

Step 7 When you have finished adding members to the extension list, click **Save** to store your changes.

To create a new pickup group and associate it with an extension list:

Step 1 With ShoreWare Director still open, click on the **Call Control** link.

Step 2 Click on the **Pickup Groups** link to display a window similar to the one shown in Figure 9-12:

<input type="checkbox"/>	Name	Extension	Site	Switch	Members
<input type="checkbox"/>	Interflow Pickup Group	888-55403	Headquarters	108-Fuji8-hq	9

Figure 9-12 Creating a new pickup group

Step 3 Click the **New** button to display a window similar to the one shown in Figure 9-13:

Figure 9-13 Select the New Extension List

Step 4 Enter a name for the new Pickup Group in the **Name** field.

- Step 5** The **Extension** field will auto-populate.
- Step 6** Click on the **Switch** drop-down menu and select the appropriate switch for this group. You should select the switch that is physically closest to the members of the group.
- Step 7** Click the **Extension List** drop-down menu and select the extension list that you just created in the previous task.
- Step 8** Click **Save** to store your changes.

9.7 Route Points

Route points allow third-party applications complete access to call control signalling (using TAPI) and the actual voice media stream (using TAPI and WAV APIs). Configuring a route point enables calls to be terminated and controlled by a server on the network. To configure a route point, click the **Route Points** link under **Call Control** in the navigation frame. The **Route Points** list page is shown in Figure 9-14.

Route Points		Schedules		
Name	Extension	On-Hours	Holiday	Custom
CC-IVR1	4821			
CC-IVR2	4822			
CC-IVR3	4823			
CC-IVR4	4824			
CC-IVR5	4825			
Sales-IRN	4820			
Support-IRN	4900			

Figure 9-14 Route Points List panel

Then click **Add new** to invoke the **Route Point** edit page (shown in Figure 9-15).

9.7.1 Parameters

The parameters that appear on the **Route Point** edit page are as follows:

- **Name:** This is the name of the route point.
- **Extension:** This is the extension number of the route point. Each route point extension must be unique. This is a required field.

If you change the existing extension number to a new number and there is an associated mailbox, messages will be retained.

- **DID:** You can assign one DID number to a route point.
Check the first box to select DID. Make a selection from the drop-down list of area codes. Finish by specifying a range of DID numbers. This is an optional field.
- **DNIS:** The **Edit DNIS Map** button invokes the **Select DNIS Trunk Group** dialog box. This lets you select a trunk group for DNIS routing. Only trunk groups that are configured for DNIS will be presented in the dialog box. You can assign multiple DNIS numbers to a workgroup. This is an optional field.

DNIS is typically used to route 800-number calls to a workgroup or application.

Route Points New Copy Save Delete Reset Help

Edit Route Point modify new

Edit this record [Refresh this page](#)

Name:

Extension:

DID: (DID Range: +14082415301 - 5310)

DNIS:

Language:

User Group:

Route Point Server:

Mailbox (server) [Voice Mail Delivery and Notification](#)

Accept Broadcast Messages

Include in System Dial By Name Directory

Make Number Private

Fax Redirect

Call Stack Depth:

Recorded Name: no audio input

Voice Mail Password: Confirm:

On-Hours
 Off-Hours
 Holiday
 Custom

Schedule: [Edit this schedule](#)

Call Handling:

Call Forward: Always No Answer/Busy Never

Always: Extension:

External: (e.g. 9+1 (408) 331-3300)

Busy: Extension:

External: (e.g. 9+1 (408) 331-3300)

No Answer: Extension:

External: (e.g. 9+1 (408) 331-3300)

No Answer Number of Rings:

Mailbox:

Greeting: no audio input

Assistant:

Enable Calling Message Notification

Figure 9-15 Route Point Edit panel

- **Language:** Select the route point language from the drop-down list.
- **User Group:** This drop-down list lets you assign a user group to the route point.

The route point requires permissions just like a user. For instance, the route point call to external call forwarding needs access to trunk groups and has a mailbox. This is a required field.

Consult Section 10.3 on page 185.

- **Route Point Server:** This selects the server that provides route point services for third-party applications. Third-party applications gain control of calls handled by the ShoreTel 6.1 system through route points. This is a required field.

It is recommended that the route point server be a separate server from the headquarters server and not be configured with mailboxes.

Consult Section 4.4 on page 48.

- **Mailbox (server):** This provides the route point with a mailbox on the associated server. If you change the server, all messages are automatically moved to the new server. The default mailbox server is the headquarters server. This is a required field.
- **Accept Broadcast Messages:** This check box enables the route point mailbox to receive broadcast messages. This is an optional field.
- **Include in System Dial by Name Directory:** This check box includes the route point in the auto-attendant dial-by-name directory. This is an optional field.
- **Make Number Private:** This check box makes the route point extension private. When the route point is private, it is not listed in the system directory and does not appear in the Call Manager dialing lists. This is an optional field.

See also Section 10.4 on page 187.

- **Fax Redirect:** This check box enables fax redirection. When the route point answers a call and a fax tone is detected, the fax is redirected away from the route point to the headquarters fax extension. This is an optional field.

See also Section 3.2 on page 39.

- **Call Stack Depth:** This lets you specify the maximum number of simultaneous calls that can be “stacked” on the route point extension. When this number is met, additional inbound calls will be routed to the call forward busy destination.

Valid entries are 1 through 200.

See also Section 10.2.1 on page 175.

Call stack depth is a licensed feature. For more information, see Section 2.9 on page 25.

- **Recorded Name:** The *Record*, *Play*, *Enter*, and *Import* buttons let you record a name for the route point. The *Recorded Name* is used as part of the default mailbox greeting as well as in the dial-by-name directory. This is an optional field.

You can use your PC microphone and speakers or a telephone to play and record within ShoreWare Director. Please refer to the auto-attendant options for more information.

You can also import prompts into ShoreWare Director. Prompts must be recorded as μ-law WAV files.

- **Voice Mail Password:** This is the password used for accessing a route point voice mailbox over the telephone. Enter a password in the first text-entry field and again in the *Confirm* field. This is a required field.

The default password is “1234”. Passwords may be numeric only.

- **Schedule:** You can configure schedules for the *On-Hours*, *Holiday*, and *Custom* modes that automatically change the call handling of the route point. The rules for schedules are:

- If it is custom time, use *Custom* mode.
- If it is holiday time, use *Holiday* mode.
- If it is on-hours time, use *On-Hours* mode.
- Otherwise, use *Off-Hours* mode.

If *no schedules* are specified, *On-Hours* mode is used.

The *Edit this schedule* link provides a quick way to navigate to the associated schedule. This is an optional field.

- **Call Handling Call Forward:** These buttons let you specify when calls are forwarded. The conditions are *Always*, *No Answer/Busy*, and *Never*.
 - *Always*—The Always condition forwards calls to the number specified in the Always Destination parameter immediately when a call is received.
 - *Busy*—The Busy condition forwards calls to the Busy Destination immediately if the user's call stack is full.
 - *No Answer*—This condition forwards calls if there is no answer.
 - *No Answer Number of Rings*—Sets the number of rings after which a no answer condition is assumed to exist.

This is a required field.

- *Always Destination*—When the *Always* call forward condition is selected, calls are forwarded immediately to this extension. You can also forward calls to an external number (access code required).
 - *Busy Destination*—When the *No Answer/Busy* call forward condition is selected, calls are forwarded to this extension immediately if the user's call stack is full. You can also forward calls to an external number (access code required).
 - *No Answer Destination*—When the *No Answer/Busy* call forward condition is selected, calls are forwarded to this extension after the specified number of rings. You can also forward calls to an external number (access code required).
- **Mailbox Greeting:** This lets the user record a greeting for his or her mailbox, using the *Record*, *Play*, *Erase*, and *Import* buttons. This is on by default.
 - **Assistant:** Each route point can have a personal assistant, which is the destination a calling party is transferred upon dialing "0" in the route point mailbox. This is an optional field.
 - **Enable Calling Message Notification:** This check box enables message notification for this call handling mode. The manner in which the user is notified is determined by the user's message notification settings. The recommended default is off.

9.8 Call Control Options

Set the general call control options from the Call Control Options page, as shown in Figure 9-16.

9.8.1 Distributed Routing Service

Distributed Routing Service (DRS) allows larger systems to scale beyond 60 switches to 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 Distributed Routing Service is disabled, ShoreGear switches in a system 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 calls are placed from any extension, each switch is able to route the call to the correct ShoreGear switch based on its internal routing database.

Call Control Options

Edit

Edit this record

Refresh this page

General:

Use Distributed Routing Service for call routing.

Enable Monitor / Record Warning Tone.

Generate an event when a trunk is in-use for minutes.

Park Timeout (1-100000) after seconds.

Hang up Make Me Conference after minutes of silence.

Delay before sending DTMF to Fax Server: msec

Enable SIP Session Timer.

Session Interval (90 - 3600): sec

Refresher:

Voice Encoding and Quality of Service:

Intra-Site Calls:

Inter-Site Calls:

FAX and Modem Calls:

Maximum Inter-Site Jitter Buffer: msec

DiffServ / ToS Byte (0-255):

Admission control algorithm assumes RTP header compression is being used.

Enable Media Encryption.

Always Use Port 5004 for RTP.

Trunk-to-Trunk Transfer and Tandem Trunks:

Hang up after minutes of silence.

Hang up after minutes.

Figure 9-16 Call Control Options Edit panel

When Distributed Routing Service is enabled, ShoreGear switches only exchange routing information with other switches at the same site rather than exchanging routing information with every other switch in a multi-site system. Although each 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 needed 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 and switches first try to contact the nearest instance of the Distributed Routing Service in the hierarchy. If that instance of DRS is not reachable, the instance of DRS at the parent server in the hierarchy will be contacted as a fallback. If

both instances of DRS are not reachable the switch will make a best effort to route the call based on its internal routing tables built from communicating with peer ShoreGear switches at the same site.

9.8.2 Parameters

The parameters on the *Call Control Options* edit page are as follows:

9.8.2.1 General

- **Use Distributed Routing Service for call routing:** Enables Distributed Routing Service (see preceding explanation.)
- **Enable Monitor / Record Warning Tone:** When a 2-way or Make Me Conference call is monitored or recorded, checking this box causes a tone to be played that warns of monitoring or recording. This option is enabled by default.

Deselect this check box to enable silent recording. Silent recording allows operators and supervisors to hide the fact that they are recording agents' calls by “silently” recording those calls. This behavior may be desirable in certain situations, such as monitoring the telephone manners of an employee.

When the recording is silent or hidden, PCM offers no visual or audible indication that the call is being recorded. The periodic beeping sound (used to notify call participants that their calls are being recorded) is suppressed.

Also refer to Section 10.2.1 on page 175 and Section 10.4.5 on page 199 for information that impacts recording and monitoring.

Example: ShoreTel does not warrant or represent that your use of call monitoring or recording features of the Software will be in compliance with local, state, federal or international laws that you may be subject to. ShoreTel, Inc. is not responsible for ensuring your compliance with all applicable laws.

Before disabling the warning tone, you may wish to consult with legal counsel regarding your intended use.

- **Generate an event when a trunk is in use for N minutes:** You can set this parameter to generate an event log when a trunk has been in use for the specified time period.
- **Park Timeout after NNNNNN seconds:** You can set how long a call will remain on park before the call returns to the party that parked the call. The timeout is in seconds and can have a value from 1 to 100,000 seconds. Unchecking the *Park Timeout* check box allows calls to be parked indefinitely.
- **Hang Up Make Me Conference after NN minutes of silence:** The default timeout is 20 minutes. If a conference is silent for the set length of time, a hang-up will be forced.
- **Delay before sending DTMF to Fax Server:** Enter the amount of delay (in milliseconds) after which the DTMF information will be sent to the fax server. Consult your fax server documentation to find out how much delay the fax server expects.
- **Enable SIP Session Timer:** Select this check box to have keepalive heartbeats sent to SIP endpoints.
- **Session Interval (90-3600):** Enter the number of seconds in the **Session Interval** field to specify the keepalive interval at which heartbeats will be broadcast. The heartbeat will be sent out at the specified period and if no response is received, the session will be dropped. (See RFC 4028 for details on this parameter.)

- **Refresher:** Use the **Refresher** drop-down menu to specify whether the SIP Session Timer will be applied to the caller or callee. Choices are None, Caller UAC, and Caller UAS. (See RFC 4028 for details on this parameter.)

9.8.2.2 Voice Encoding and Quality of Service

- **Intra-Site Calls (calls within a site):** This drop-down list lets you select the type of encoding used when making calls within your site. Choose one of these encodings:
 - Linear Broadband (256 Kbps)
 - Linear (128 Kbps)
 - G.711 (64 Kbps)—the default
 - ADPCM (32 Kbps)
 - G.729a (8 Kbps)
- **Inter-Site Calls (calls between sites):** This drop-down list lets you select the type of encoding used when making calls between your sites. Choose one of these encodings:
 - Linear Broadband (256 Kbps)
 - Linear (128 Kbps)
 - G.711 (64 Kbps)
 - ADPCM (32 Kbps)—the default
 - G.729a (8 Kbps)
- **FAX and Modem Calls:** This drop-down list lets you select the type of encoding used for faxing or when calls are made from a modem. Choose one of these encodings:
 - Linear (128 Kbps)
 - G.711 (64 Kbps)—the default
- **Maximum Inter-Site Jitter Buffer:** This parameter sets the maximum size of the jitter buffer. A larger jitter buffer might result in more delay between calling parties, which might degrade the quality of service. The buffer can be set from 20 to 300 msec. The default is 50 msec.
- **DiffServ/TOS Byte:** Enter a Types of Service (TOS) value or a DiffServ value, whichever your network currently supports. This byte is applied to media originating from all ShoreGear physical switches (SoftSwitches excluded) and IP phones. The value entered in this field must be a decimal number 0 to 255.
- **Admission control algorithm assumes RTP header compression is being used:** To enable this feature, select the check box.
- **Enable Media Encryption:** To enable this feature, select the check box. Note that encryption occurs at the application layer and that only calls inside a ShoreTel 6.1 network will be encrypted. Once the call passes through TDM filtering the encryption is stripped away and the conversation is no longer encrypted.
- **Always Use Port 5004 for RTP:** By default, this box is checked. You should clear this check box to enable interoperability with third-party SIP devices.

All IP Phones and switches must be rebooted for settings to take effect. Failure to reboot may result in one way media.

9.8.2.3 Trunk-to-Trunk Transfer and Tandem Trunks

This lets you manage trunk-to-trunk transfers when the *Allow Trunk-to-Trunk Transfer* parameter is enabled on the *Telephone Features Permissions* class of service edit page. The field is defined in the section called Section 10.2.1 on page 175.

The ShoreTel 6.1 system supports trunk-to-trunk transfers, which allow a user to transfer an external caller to an external number. Since this feature can lead to unwanted toll charges, the system also supports a class of service permission that only grants this feature to selected user groups. To grant this permission, you enable the *Allow Trunk-to-Trunk Transfer* parameter on the *Telephone Features Permissions* class of service edit page.

Users with trunk-to-trunk transfer permission might accidentally transfer an external caller to an external number without realizing it. This can lead to “hung” trunks, resulting in the inability to make outbound calls or take inbound calls.

The ShoreTel 6.1 system lets you manage trunk-to-trunk transfers using the *Call Control Options* page. This lets you eliminate unwanted trunk-to-trunk transfers while ensuring that valid trunk-to-trunk transfers are not dropped.

The following are considered trunk-to-trunk transfers:

- A user is talking with an external party and transfers the external party *blindly* or *consultatively* to an external number.
- A three-party conference call is taking place with one user and two external parties, and the user drops from the call.

An external party forwarded to an external number by a user’s call handling mode is not a trunk-to-trunk transfer.

- **Hang up after N minute(s) of silence:** Enabling this parameter automatically drops trunk-to-trunk transfers after both parties have been silent for the defined time period. The default time period is 60 minutes.
- **Hang up after N minutes:** Enabling this parameter automatically hangs up the trunk-to-trunk transfer after the defined time period. *This parameter should be set only as a last resort, and set for a long time period.* The default time period is 480 minutes.

9.9 Bandwidth Management and Codec Negotiation

ShoreWare supports additional codecs available on ShorePhone and through SIP devices. Codec negotiation during voice call setup is facilitated by data structures, including codec lists and profiles.

Bandwidth Management and Codec Negotiation tools available through Director include:

- Codec lists that are configurable through Director.
- Video codec support
- A Director parameter that permits inter-site video sessions.
- SIP codec negotiation method that complies with RFC 3264 – An Offer / Answer Model with the SDP.

9.9.1 Codec Lists

Codec lists enumerate a set of codecs. Director defines two types of codec lists: Supported Codecs and Preferred Codecs.

9.9.1.1 Supported Codecs

Supported Codecs is a comprehensive list of all codecs available to system devices. When ShoreWare is initially installed, the Supported Codecs list comprises the set of codecs provided by ShoreWare and available on ShorePhones. Administrators can add codecs to support SIP devices that may use codecs not initially provided by ShoreWare.

The Supported Codecs list also indicates the bandwidth required by each codec. The bandwidth numbers are used by ShoreWare to allocate bandwidth as voice calls are initiated and terminated.

Supported Codecs list contents, including the bandwidth settings, are passed to all switches in the system, where they are used for selecting codecs for individual call sessions.

9.9.1.2 Preferred Codec Lists

Preferred Codec lists are subsets of the Supported Codecs list. Preferred Codec lists are referenced by Sites within a ShoreWare system to specify the codecs used for intersite and intrasite voice calls. ShoreWare provides six default codec lists which cannot be deleted or modified. Administrators can define additional lists through Director.

The default codec lists provided with ShoreWare include:

- Fax Codecs
- High Bandwidth Codecs
- Low Bandwidth Codecs
- Medium Bandwidth Codecs
- Very High Bandwidth Codecs
- Very Low Bandwidth Codecs

9.9.2 Intersite Video Sessions

Intersite Video Sessions are enabled through a Telephony Class of Service setting. ShoreWare does not allocate bandwidth for video calls.

9.9.3 Codec Negotiation

ShoreWare Version 8.1 supports simultaneous audio, video, and data codec negotiations to facilitate multimedia sessions between SIP endpoints as defined by RFC 3264 – An Offer / Answer Model with the SDP. Codecs specified in the Supported Codec list are offered during session parameter negotiations.

The ShoreWare negotiation process that supports RFC 3264 is as follows:

1. The calling device sends the list of codecs it supports to the ShoreGear servicing the call.
2. The ShoreGear switch that controls the calling device compiles a codec list. The codec list contains all codecs contained in the following:
 - The calling device's codec list
 - One of the site's codec lists – intersite, intrasite, or fax – depending on the call typeCodecs on the combined list are sorted as specified by the selected site codec list.
3. (*Intersite Calls only*): The ShoreGear switch that controls the destination device modifies the codec list by removing all codecs that are not listed on the destination site's codec list.
4. The ShoreGear switch controlling the destination device sends the codec list to the destination device.

5. The destination device replies by sending a list of one or more codecs to the originating device.
6. The list sent by the destination device typically includes, from the received codec list, the highest priority codec it can support.
7. The two devices begin sending RTP streams using the highest priority codec listed in the destination device's reply.

9.9.4 Implementation

9.9.4.1 Supported Codecs panel

The Supported Codecs panel lists the audio codecs that are available to devices making voice calls through ShoreWare. Codec lists used when negotiating call parameters comprise codecs displayed on this panel. Although most commonly used codecs are listed on this panel, the system can also support other codecs by adding them to the list.

To access the Supported Codecs panel, shown in Figure 9-17, select *Administration -> Call Control -> Supported Codecs* from the Director menu. Each row in the table lists a codec.

<input type="checkbox"/>	Name	Bandwidth (in kbps)	Default
<input type="checkbox"/>	AAC_1C/32000	96	Yes
<input type="checkbox"/>	BV16/8000	16	Yes
<input type="checkbox"/>	BV32/16000	32	Yes
<input type="checkbox"/>	DV14/8000	32	Yes
<input type="checkbox"/>	G722/8000	64	Yes
<input type="checkbox"/>	G729/8000	8	Yes
<input type="checkbox"/>	L16/16000	256	Yes
<input type="checkbox"/>	L16/8000	128	Yes
<input type="checkbox"/>	PCMA/8000	64	Yes
<input type="checkbox"/>	PCMU/8000	64	Yes

Figure 9-17 Supported Codecs panel

Table fields in the Supported Codecs panel include:

- **Name:** This parameter is fully qualified codec ID string of the codec. ShoreWare uses this string to specify codecs while negotiating with other calling devices.
The codec ID string comprises the name and sampling rate of the codec. While the codec name generally corresponds with the name by which the codec is commonly referred, PCMA specifies a G.711 codec (A-law) and PCMU specifies a G.711 codec (μ -law).
- **Bandwidth (in kbps):** This parameter identifies the bandwidth required by the codec. ShoreWare uses this figure when allocating bandwidth resources.
- **Default:** This parameter specifies the source of the codec entry:
 - *Yes* indicates a codec that was provided with ShoreWare. Default codecs cannot be removed from the list.
 - *No* indicates a codec that was provided by the system administrator. These codecs can be removed from the list.

To add a new codec to the list,

Step 1 Click the Add button at the top of the panel to open the Supported Codec Info popup, shown in Figure 9-18.

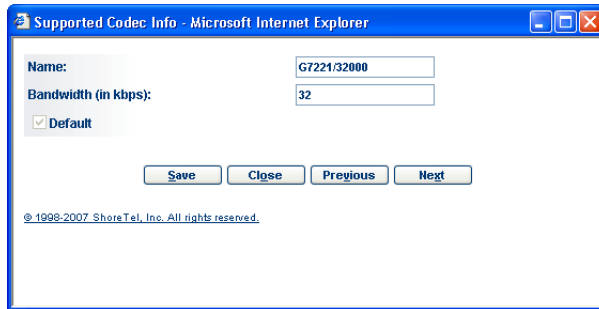


Figure 9-18 Supported Codec Info popup

Step 2 Enter the fully qualified codec ID string in the Name data entry field.

The field must be entered exactly as expected by devices that negotiate call parameters.

Step 3 Enter the codec bandwidth in the Bandwidth data entry field.

Entering incorrect numbers in this field compromises ShoreWare's ability to manage bandwidth resources.

Step 4 Press the Save button to store the changes to the database and return to the Supported Codecs panel.

The Default field is used when editing existing codecs to denote a codec that is supplied with ShoreWare. This parameter is not editable.

To edit an existing codec:

Step 1 Click the name of the codec to be edited to open the Supported Codec Info popup, shown in Figure 9-18.

Step 2 Change the contents of the Name or Bandwidth data entry fields.

The Default box is not editable.

Step 3 Press SAVE to store the changes and return to the Supported Codecs panel.

9.9.4.2 Codec Lists

Codec lists are subsets of the codecs supported by the ShoreWare system. Codec lists are selected in the Edit Sites panel to determine the codecs used for intersite and intrasite calls. Director panels that support codec lists include the Codec List panel and the Edit Codec Lists panel.

Each codec list must include the PCMU (G.711 μ -law) codec.

Codec List panel

The Codec List panel, shown in Figure 9-19, provides a roster of Codec Lists configured on the system. Codec list names, as specified on this panel, are used on the Edit Sites panel when selecting codecs for intersite and intrasite calls. To access the Codec List panel, select *Administration -> Call Control -> Codec Lists* from the Director menu. Each line in the table corresponds to a Codec List.

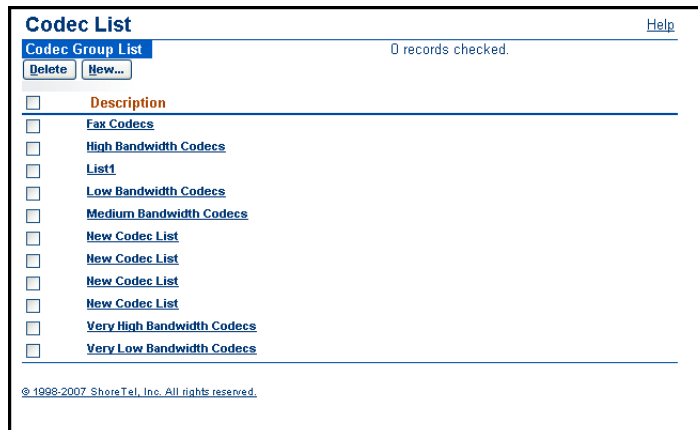


Figure 9-19 Codec List panel

To add a new codec list, open the Edit Codecs Lists panel by clicking the New button in the top left corner of the panel.

To view or edit an existing codec list, open the Edit Codec List panel by clicking the name of the desired codec list. Default codec lists supplied with ShoreWare cannot be edited.

To delete an existing codec list, select the box left of the desired codec and press the Delete button in the top left corner of the panel. Default codec lists supplied with ShoreWare cannot be deleted.

Edit Codec List

The Edit Codec List panel, shown in Figure 9-20, modifies the Codec List specified by the Name field contents. The panel displays two tables:

- The **Choose Codec** table lists the codecs available on the system that are not in the specified codec list.
- The **Codec List Members** table displays the codecs that are in the specified codec list.

All codecs available on the system, as defined by the Supported Codecs panel, are listed either in the Choose Codec table or the Codec List Members table.

To add a codec to the codec list, select the desired codec in the Choose Codec, then click the ADD button between the tables.

To remove a codec from the codec list, select the desired codec in the Codec List Members table, then click the REMOVE button between the tables.

To change the position of a codec within the Codec List Members table, select the desired codec in that table and click the MOVE UP or MOVE DOWN buttons located between the tables.

To save Codec List changes, click the SAVE button at the top of the panel

To create a new codec list, press the NEW button at the top of the panel.

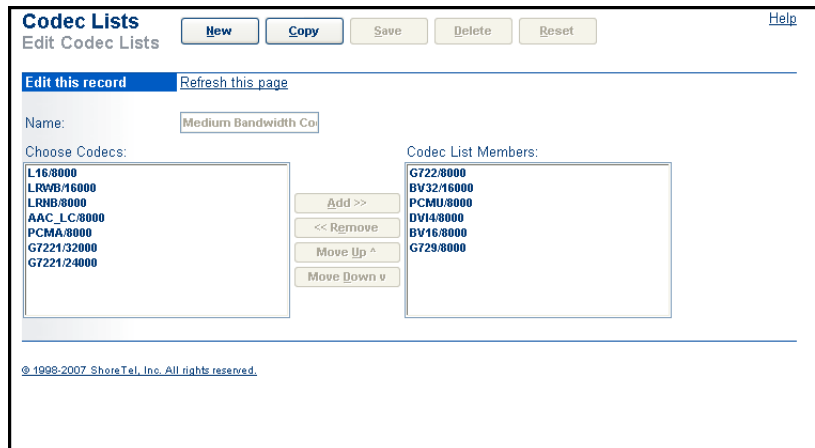


Figure 9-20 Edit Codec List panel

To use the displayed codec list as a template for a new list, click the *COPY* button at the top of the panel.

To revert the codec list to the last saved version, click the *Reset* button at the top of the panel.

Default Codec Lists

ShoreWare provides six default codec lists that cannot be modified or deleted. The following enumerates the codecs, in order of priority, within each list:

- **Very High Bandwidth Codecs**
 - L16/16000
 - G722/8000
 - BV32/16000
 - L16/8000
 - PCMU/8000
 - DVI4/8000
 - BV16/8000
 - G729/8000
- **High Bandwidth Codecs**
 - G722/8000
 - BV32/16000
 - L16/8000
 - PCMU/8000
 - DVI4/8000
 - BV16/8000
 - G729/8000
- **Medium Bandwidth Codecs**
 - G722/8000
 - BV32/16000
 - PCMU/8000
 - DVI4/8000
 - BV/8000
 - G729/8000

- Low Bandwidth Codecs
 - BV32/16000
 - DVI4/8000
 - BV16/8000
 - G729/8000
 - PCMU/8000
- Very Low Bandwidth Codecs
 - G729/8000
 - PCMU/8000
- Fax Codecs
 - L16/8000
 - PCMU/8000

9.9.4.3 Intersite Video

To configure the Intersite Video setting for a Class of Service:

- Step 1** Access the Class of Service list panel by selecting *Administration -> Users -> Class of Service* from the Director menu.
- Step 2** Click the name of the desired Telephony Features Permissions Class of Service in the top table of the panel.
- Step 3** Select *Allow Intersite Video Calls* to allow members of the Class of Service to participate in video calls. Clear this setting to deny access to video calls.

Figure 9-21 indicates the location of the *Allow Intersite Video Calls* checkbox.

The screenshot shows the 'Class of Service' configuration page for 'Fully Featured'. The page includes a header with 'Edit Telephony Features Permissions' and buttons for 'New', 'Copy', 'Save', 'Delete', 'Reset', and 'Help'. Below the header is a table with one row for 'Fully Featured'. The table has columns for 'Name', 'Max. Call Stack Depth', 'Max. Buddies Per User', 'Max. Parties in Make Me Conference', and 'Allow Intersite Video Calls'. The 'Allow Intersite Video Calls' checkbox is currently unchecked and is highlighted with a red arrow. The footer of the page contains the copyright notice: '@ 1998-2007 ShoreTel, Inc. All rights reserved.'

Figure 9-21 Allow Intersite Video Calls parameter

9.10 Automatic Ringdown

An **automatic ringdown** circuit comprises predefined devices at the circuit endpoints and is configured to ring a recipient device immediately after an initiating device goes off hook. Automatic ringdown calls are completed without dialing or any other signaling other than the initiating device going off hook.

The simplest example of an automatic ringdown circuit consists of two points, with one telephone at each end of the circuit. When the telephone at one end goes off-hook, the phone at the other end immediately rings. Phones without dials may be used on each end of the circuit.

Ringdown circuits may be unidirectional or bidirectional.

- In unidirectional circuits, calls always originate from the same end of the network. A device on a receiving endpoint going off hook has no effect on the initiating endpoint device.
- In bidirectional circuits, both ends can initiate and receive automatic ringdown calls.

9.10.1 Dedicated Circuit Ringdown

ShorePhones with call appearance buttons and the ShoreWare Button Box support Dedicated Circuit Ringdown. A call appearance button is programmed for Ringdown by configuring it as a Bridged Call Appearance and selecting a Ringdown option. When using ShorePhones as the calling and recipient devices, an IP Phone button on both devices is configured for Ringdown. ShoreWare supports using external devices as the ringdown recipient.

A ringdown call is initiated by pressing the ringdown button on a calling device. The ringdown button on the recipient device blinks; the recipient device can be configured to ring or remain silent on an inbound ringdown call. The ringdown call is answered on the recipient phone either by lifting the handset or pressing the blinking ringdown button. If the phone is configured to remain silent when a ringdown call is incoming, the blinking button must be pressed to answer the call.

ShorePhones can be configured for one-way or two-way ringdown. When phones are configured for one-way ringdown, one phone is defined as the recipient device; pressing the ringdown button on that device will not initiate a call to the phone on the other end of the circuit. When phones are configured for two-way operation, pressing the ringdown button on either device initiates a call to the device on the other end of the circuit.

To force a device to ring until it is answered, the call stack depth for Bridge Call Appearances supporting ringdown should be set to 1 and call handling mode transfers should be disabled by selecting **No Answer**.

The following sections describe four typical ringdown configurations.

9.10.1.1 Basic One-to-One Ringdown

Basic one-to-one ringdown, as depicted in Figure 9-22, is enabled by configuring a Call Appearance button on each phone as a Bridged Call Appearance (BCA). During an incoming ringdown call, the BCA button on the recipient device blinks.

ShoreWare supports one-way or two-way ringdown in this configuration.

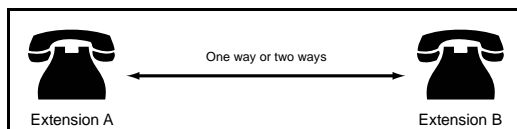


Figure 9-22 One to One Ringdown Operation

9.10.1.2 One-to-many Ringdown

ShoreWare supports one-to-many ringdown operation as depicted in Figure 9-23. When the Ringdown BCA is pressed on the calling device, Extension A in Figure 9-23, the corresponding button on all recipient devices programmed to receive the ringdown call

flashes Green and off. When the call on one of the recipient devices is answered, the Ringdown button on the other devices is Red. If the answering device places the call on hold, the call is parked to the Bridged Call Appearance and is available for all of the recipient devices. Conference calls involving a ringdown number that is placed on hold is not parked onto the BCA extension.

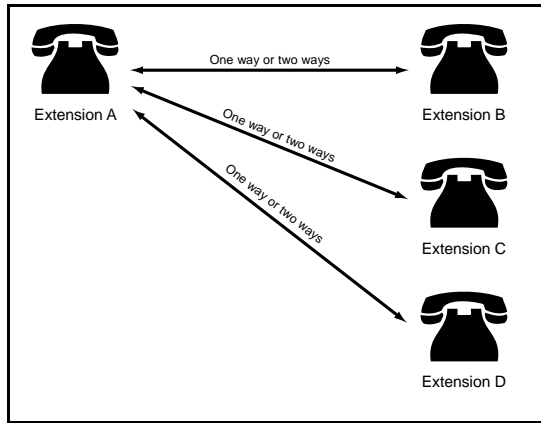


Figure 9-23 One to Many Ringdown Operation

ShoreWare supports two-way ringdown for the one-to-many configuration. When a ringdown call is placed from a recipient device, (Extension B in Figure 9-23), the BCA button on the other recipient devices (Extensions C and D) becomes solid red when Extension A answers the call, indicating that the line is busy. Pressing these red buttons has no effect. If Extension A does not answer the ringdown call, the phone rings until the caller on Extension B hangs up or the call handling mode for Extension A handles the call.

9.10.1.3 Multiple ring down buttons

ShorePhones with sufficient call appearance buttons support multiple ringdown circuits. In Figure 9-24, Extension A is configured to support two ringdown circuits:

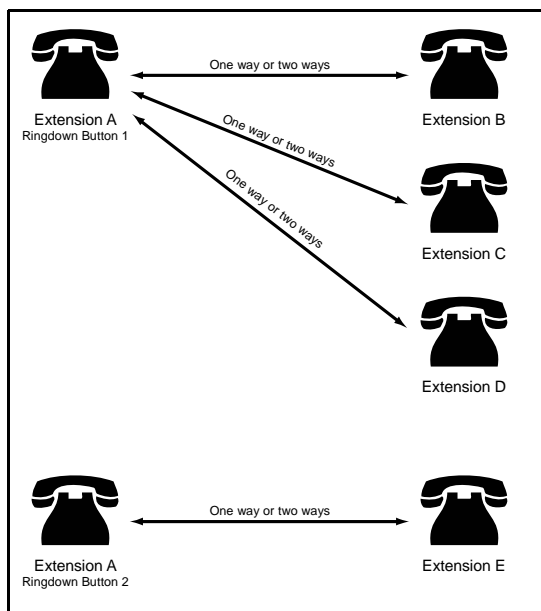


Figure 9-24 Multiple Ringdown Buttons

- one ringdown circuit with Extensions B, C, and D at the other end.
- one ringdown circuit with Extension E at the other end.

This configuration requires four Bridge Call Appearance extensions: two BCA extensions for Extension A; one BCA extension shared by Extensions B, C, and D; and one BCA extension for Extension E.

9.10.1.4 Ring Down to an External Device

Ringdown to an external device is supported by programming the ringdown button to access a trunk group that has a unique trunk access code and contains only one trunk. When the user presses the ringdown button, it accesses that trunk.

To place a call over a ringdown circuit to an external device, the user enters the trunk access code in addition to the phone number of the device. For instance, when configure a analog trunk group to service the ringdown call, the default trunk access code of 9 must be dialed to place a ringdown call.

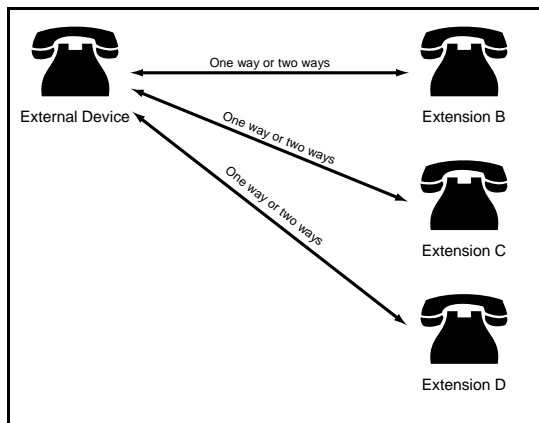


Figure 9-25 Ringdown from an External Device

This trunk is not required to be reserved solely as the ringdown circuit. Any user can dial this trunk access code to select this trunk. If the trunk is busy, pressing the ringdown button generates a busy signal. “Enable CHM” only applies for an incoming call. It is not applicable for an outbound call. To enable ringdown buttons on the ShoreWare devices (Extensions B, C and D in Figure 9-25), the BCA extension must be configured on the Trunk Groups page.

9.10.2 Phone Delayed Ringdown

ShoreWare permits IP phones that provide a dial tone to perform ringdowns by taking the handset off hook. If a number is not dialed within a specified period after taking the phone off hook, a ringdown call is directed to the predefined recipient device.

When the ringdown device goes off hook by lifting up handset or pressing speaker or headset, the ringdown number is dialed if a digit is not entered within the configured ringdown delay period. If a digit is entered within this period, the ringdown call is not performed.

Analog phones to which a user extension is assigned can receive incoming calls even when configured for delayed ringdown calls. When an anonymous user is assigned to the port, the device cannot receive incoming calls if it is configured to make delayed ringdown calls.

9.10.3 Implementation

Automatic ringdown circuits may also be configured in a one-to-many topology. In these circuits, one initiating device rings a group of phones on the other circuit endpoints.

Automatic ringdowns are activated on ShorePhones by pushing an IP Phone Call Appearance button or lifting a handset on a dedicated ringdown device. These actions cause the recipient device to continuously ring until the call is answered or the calling party ends the call. The Call Handling Mode on the recipient device is ignored for ringdown calls. Automatic ringdown is supported only by phone; Call Manager does not support automatic ringdown.

ShoreWare implements Ringdown through the following methods:

- **Dedicated Circuit Ringdown:** Ringdown is immediately initiated when a programmed Call Appearance button is pushed or a specified device goes off hook. The ringdown call is answered on the recipient by pressing the corresponding Call Appearance button or taking the device off hook.
- **Phone Delayed Ringdown:** ShoreWare permits IP phones that provide a dial tone to perform ringdowns by taking the handset off hook. If a number is not dialed within a specified period after taking the phone off hook, a ringdown call is directed to the recipient device.

9.10.3.1 Director Configuration panels

Ringdown circuits are implemented by configuring IP Phone buttons for circuit endpoints through Director. Delayed ringdown circuits are configured from the Edit User – General panel.

Configuring IP Phone Buttons

IP Phone Buttons for ShoreWare users are configured from the Program IP Phone Buttons panel, as shown in Figure 9-26. To access the Program IP Phone Buttons panel, open the Edit User – Personal Options panel for the desired user, then click Program IP Phone Buttons.

Program IP Phone Buttons Save Reset Help

Bonnie Hopkins

Edit Custom Keys Refresh this page

IP Phone Type:

#	Function	Label	Target
1	All <input type="text" value="Call Appearance"/>		
2	Telephony <input type="text" value="Bridged Call Appearance"/>	b1	
3	All <input type="text" value="Call Appearance"/>		

Extension: Search

Call Stack Position:

Ring Delay Before Alert:

Show Caller ID on Monitored Extensions:

Never Only When Ringing Always

Enable Auto-Answer When Alerting

No Connected Call Action:

Answer Only

Dial Tone

Dial Extension: Search

Dial External: (e.g. 9+1 (408) 331-3300)

© 1998-2007 ShoreTel, Inc. All rights reserved.

Figure 9-26 IP Phone Button Ringdown options

In Figure 9-26, button #2 is configured to monitor the Bridged Call Appearance specified by the Extension data entry field. Selecting **Answer Only** disables the ringdown feature on the specified button. BCA options located below the Normal Operation option configure ringdown options for the button.

BCA Ringdown parameters include:

- **Dial Tone:** Select this option to configure the phone as the recipient on a ringdown circuit. Buttons configured with Dial Tone cannot initiate ringdown calls.
- **Dial Extension:** Select this option to configure the button as the initiating end of a ringdown circuit when the recipient is an IP phone located on the ShoreWare network. The Extension data entry field specifies the Bridged Call Appearance that is dialed when the IP Phone button is pushed. Buttons configured to perform ringdown calls can also receive ringdown calls.
- **Dial External:** Select this option to configure the button as the calling end of a ringdown circuit when the recipient is a device not located on the ShoreWare network. The data entry field specifies the phone number that is dialed when the IP phone button is pushed; the trunk access code required to access the trunk group dedicated to this ringdown circuit must be included as part of the phone number.

Configuring a Delayed Ringdown Circuit

Delayed ringdown circuits are configured on the Edit User – General panel for the user extension that initiates the ringdown call. Figure 9-27 displays the section of the Edit – User General panel containing the delayed ringdown parameter fields, which are located in the bottom third of the panel.

The screenshot shows the 'Users' administration page for editing a user. The 'General' tab is selected, and the 'Delayed Ringdown' checkbox is checked. The configuration fields are as follows:

- First Name:** Adam
- Last Name:** Pierce
- Number:** 122
- Allow Mobile Call Manager
- Prompt for Password:** First Time Only Always
- Delayed Ringdown
 - Extension: [] Search
 - External Number: [] (e.g. 9+1 (408) 331-3300)
 - Ringdown Delay:** [] sec
- Client User ID:** 122

© 1998-2007 ShoreTel, Inc. All rights reserved.

Figure 9-27 Delayed Ringdown Configuration Parameters

Delayed Ringdown Circuit parameters include:

- **Delayed Ringdown:** Select this checkbox to enable Delayed Ringdown for the extension. If this parameter is not enabled, all other Ringdown parameters are not selectable.

- **Extension:** Select this option to specify a ShoreWare extension as the recipient device, then enter the extension of the recipient device in the corresponding data entry field.
- **External Number:** Select this option to specify a phone outside of the ShoreWare network as the recipient device, then enter the phone number of the recipient device, including the Trunk Access Code of the ringdown trunk, in the corresponding data entry field.
- **Ringdown Delay:** Enter the period of time that the phone waits after the user picks up the handset before initiating the ringdown call. Enter 0 in this field to cause the phone to immediately dial the ringdown device whenever the handset is picked up.

9.10.3.2 Ringdown Procedures

The following sections describe the procedures required to implement various ringdown circuit configurations.

Implementing a One-to-One Unidirectional Ringdown Circuit

Unidirectional ringdown circuits require two Bridged Call Appearances – one for the calling device and one for the recipient device. Both devices in this procedure are ShoreWare extensions.

To implement a One-to-One Unidirectional Ringdown Circuit:

Step 1 Create two Bridged Call Appearances

Bridged Call Appearances are listed in the Bridged Call Appearance list panel, which is accessed by selecting *Administration -> Call Control -> Bridged Call Appearances* from the Director menu. Press the NEW button at the top of this panel to create new BCAs. Refer to the System Administrator's Guide for information on creating Bridged Call Appearance.

Step 2 Program an IP Phone Button on the calling device to make ringdown calls, as follows:

Step a Open the Individual User list panel in Director by selecting *Administration -> Users -> Individual Users* from the main menu.

Step b Open the Edit User panel by clicking the First Name of the desired user in the Individual User List.

Step c Select *Personal Options* on the panel selection bar at the top of the panel.

Step d Click the Program IP Phone Buttons link on the Personal Options panel to open the Program IP Phone Buttons panel.

Figure 9-26 displays the Program IP Phone Buttons panel.

Step e Select *Bridged Call Appearance* as the function for the desired IP button.

Step f Enter the first Bridged Call Appearance extension number created in Step 1 in the Extension Field located right of the button function selection fields.

Step g Select *Ringdown Extension*, located below the Extension field, and enter the second Bridged Call Appearance number in the corresponding data entry field.

Step h Press the SAVE button at the top of the panel.

- Step 3** Program an IP Phone Button on the recipient device to receive ringdown calls, as follows:
- Step a** Repeat Step a through Step e (under Step 2) for the recipient user's extension.
 - Step b** Enter the second Bridged Call Appearance extension number created in Step 1 in the Extension Field located right of the button function selection fields.
 - Step c** Select *Restricted Dial Tone*, located below the Extension Field.
 - Step d** Press the SAVE button at the top of the panel.

Implementing a One-to-One Bidirectional Ringdown Circuit

Creating a bidirectional ringdown circuits differs from creating a unidirectional circuit in that the recipient device is configured to make ringdown calls. Both devices in this procedure are ShoreWare extensions.

To implement a One-to-One Bidirectional Ringdown Circuit:

Step 1 Create two Bridged Call Appearances

Bridged Call Appearances are listed in the Bridged Call Appearance list panel, which is accessed by selecting *Administration -> Call Control -> Bridged Call Appearances* from the Director menu. Press the New button at the top of this panel to create new BCAs. Refer to the System Administrator's Guide for information on creating Bridged Call Appearance.

Step 2 Program an IP Phone Button on the calling device to make ringdown calls, as follows:

- Step a** Open the Individual User list panel in Director by selecting *Administration -> Users -> Individual Users* from the main menu.
- Step b** Open the Edit User panel by clicking the First Name of the desired user in the Individual User List.
- Step c** Select *Personal Options* on the panel selection bar at the top of the panel.
- Step d** Click the Program IP Phone Buttons link on the Personal Options panel to open the Program IP Phone Buttons panel.

Figure 9-26 displays the Program IP Phone Buttons panel.

- Step e** Select *Bridged Call Appearance* as the function for the desired IP button.
- Step f** Enter the first Bridged Call Appearance extension number created in Step 1 in the Extension Field located right of the button function selection fields.
- Step g** Select *Ringdown Extension*, located below the Extension field, and enter the second Bridged Call Appearance number created in Step 1 in the corresponding data entry field.
- Step h** Press the SAVE button at the top of the panel.

- Step 3** Program an IP Phone Button on the recipient device to make ringdown calls, as follows:
- Step a** Repeat Step a through Step e (under Step 2) for the recipient user's extension.
 - Step b** Enter the second Bridged Call Appearance extension number created in Step 1 in the Extension Field located right of the button function selection fields.
 - Step c** Select *Ringdown Extension*, located below the Extension field, and enter the first Bridged Call Appearance number created in Step 1 in the corresponding data entry field.
 - Step d** Press the SAVE button at the top of the panel.

Creating One-to-Many Ringdown Circuits

The process for creating a One-to-Many Ringdown Circuit differs from creating a One-to-One circuit in that the process of configuring the recipient devices is repeated for each recipient device in the ringdown network.

Implementing a Delayed Ringdown Circuit

A unidirectional delayed ringdown circuit requires one Bridged Call Appearance that is assigned to the recipient end of the circuit.

To create a unidirectional delayed ringdown circuit:

- Step 1** Create one Bridged Call Appearance.
- Step 2** Program a phone to make ringdown calls, as follows:
- Step a** Open the Individual User list panel in Director by selecting *Administration -> Users -> Individual Users* from the main menu.
 - Step b** Open the Edit User panel by clicking the First Name of the desired user in the Individual User List.
 - Step c** Select *General* on the panel selection bar at the top of the panel.
 - Step d** Select *Delayed Extension*.

Figure 9-27 shows the *Delayed Extension* parameter on the Edit User panel.

- Step e** Select the *Extension* parameter and enter the Bridged Call Appearance extension created in Step 1 in the corresponding data entry field.

- Step f** Enter the delay period in the *Ringdown Delay* data entry field.

The delay period is the time between when the handset is lifted and when the ringdown call is initiated.

- Step 3** Program an IP Phone Button on the recipient device to receive ringdown calls, as follows:
- Step a** Open the Individual User list panel in Director by selecting *Administration* -> *Users* -> *Individual Users* from the main menu.
 - Step b** Open the *Edit User* panel by clicking the First Name of the desired user in the Individual User List.
 - Step c** Select *Personal Options* on the panel selection bar at the top of the panel.
 - Step d** Click the *Program IP Phone Buttons* link on the Personal Options panel to open the Program IP Phone Buttons panel.
Figure 9-26 displays the Program IP Phone Buttons link.
 - Step e** Select *Bridged Call Appearance* as the function for the desired IP button.
 - Step f** Enter the initiator's extension number in the Extension Field located right of the button function selection fields.
 - Step g** Select *Restricted Dial Tone*, located below the Extension Field.
 - Step h** Press the SAVE button at the top of the panel.

Creating a Ringdown Circuit with an External Endpoint

The process for creating a ringdown circuit to an external endpoint differs from the procedure for circuits with internal endpoint as follows:

- The number of the recipient device is entered in External Number data entry fields instead of the Extension data fields.
These fields are located in the Program IP Phone Buttons panel (Figure 9-26) and the Edit User panel (Figure 9-27).
- The external number must be accessed through a specific trunk that is configured as the only trunk within a trunk group.
- The number of the recipient device includes the Trunk Access Code of the specified trunk group.

9.11 Media Encryption

ShoreWare encrypts RTP (payload) packets within the ShoreWare network. Call control packets are not encrypted.

Encryption is enabled or disabled through ShoreWare Director by a system administrator. Encryption is enabled or disabled only on a system basis and cannot be enabled for individual devices or selected calls. Encryption is transparent to the end user and end users have no control over which calls are encrypted. Changing the system encryption setting does not alter calls that are in progress; unencrypted calls in place when encryption is enabled remain unencrypted until the calls are terminated.

9.11.1 Description

ShoreWare encrypts media streams through a proprietary algorithm that uses 128-bit keys. Devices that support 128-bit encryption include

- Switches

- ShoreGear 40
- ShoreGear 60
- ShoreGear 120
- ShoreGear 220T1
- ShoreGear 220T1A
- ShoreGear 220E1
- ShoreGear 24A
- ShoreGear 90
- ShoreGear 90BRI
- ShoreGear 50
- ShoreGear 30
- ShoreGear E1
- ShoreGear T1
- ShoreGear T1k
- IP Phones
 - IP110
 - IP115
 - IP210
 - IP212k
 - IP230
 - IP265
 - IP530
 - IP560
 - IP560g
 - IP565g
 - IP8000
 - Button Box BB24

9.11.2 Implementation

The *Enable Media Encryption* parameter on the Call Control Options panel, controls encryption. To access this panel, as shown in Figure 9-28, select *Administration -> Call Control -> Options* from ShoreWare Director.

Call Control Options

Edit Save Reset [Help](#)

[Edit this record](#) [Refresh this page](#)

General:

Use Distributed Routing Service for call routing.

Enable Monitor / Record Warning Tone.

Voice Encoding and Quality of Service:

Maximum Inter-Site Jitter Buffer: msec

DiffServ / ToS Byte (0-255):

Admission control algorithm assumes RTP header compression is being used.

Enable Media Encryption.

Trunk-to-Trunk Transfer and Tandem Trunks:

Hang up after minutes of silence.

Hang up after minutes.

© 1998-2007 ShoreTel, Inc. All rights reserved.

Figure 9-28 Enable Media Encryption option on the Call Control Options panel

Configuring Users

This chapter provides information about configuring user parameters. The sections in this chapter are:

- Classes of Service
- User Groups
- Individual Users
- Notify Users
- Anonymous Telephones
- Extension Lists
- Batch Update Utility
- Call Handling

10.1 Overview

The *Users* link in the navigation frame, shown in Figure 10-1, gives you access to the *Individual Users*, *User Groups*, *Class of Service*, *Notify Users*, *Anonymous Telephones*, *Extension Lists*, *Batch Update Utility*, and *Call Handling Mode Defaults* list and edit pages. From these pages, you can add and edit users as well as assign them properties within the ShoreTel 6.1 system.

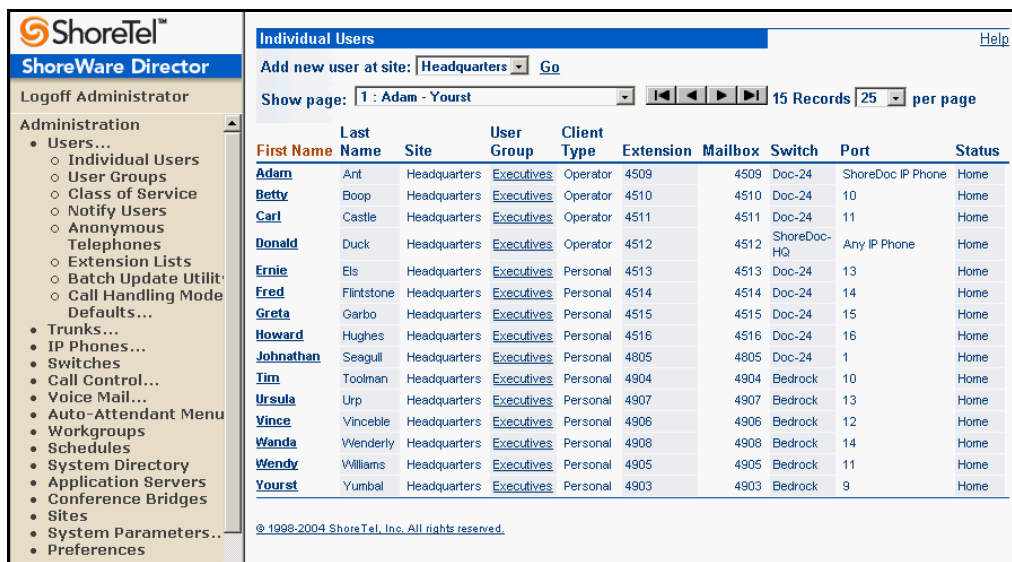


Figure 10-1 Users Navigation Links

Configuring users requires that you configure class of service first, user groups second, and individual users last. Having the following information at hand will expedite the user configuration process:

- A list or outline of the classes of service that will be associated with your user groups
- A list or outline of user groups that will be associated with individual users
- A list of new users who will be added to the system

The ShoreTel 6.1 system comes with default classes of service and user groups.

10.2 Classes of Service

Configuring classes of service is the first step in configuring users. Expand the *Users* link and click *Class of Service*. From the *Class of Service* page, shown in Figure 10-2, you configure permissions for telephony features, outside calling, and voice mail usage. Descriptions and instructions are provided for these permissions in the following sections:

- Telephony Features Permissions
- Call Permissions
- Voice Mail Permissions

Class of Service Help	
Telephony Features Permissions Add new	Name Fully Featured Minimally Featured Partially Featured Voice Mail Featured
Call Permissions Add new	Name Internal Only Local Only Long Distance Calling No Restrictions
Voice Mail Permissions Add new	Name Large Mail Box Medium Mail Box No Mail Box Small Mail Box

Figure 10-2 Class of Service Page

Each *Class of Service* type (CoS Telephony Permissions, Call Permissions, Voice Mail Permissions) has an *Add new* link and a list of names associated with that Class of Service type. To add a new CoS of a specific type (CoS Telephony, for example), click the *Add new* link to display the Telephony Features edit page, shown in Figure 10-3.

If you would like to edit an existing CoS, click on the name of the desired CoS from the list of names on the right side of the Class of Service page.

10.2.1 Telephony Features Permissions

This section describes the classes of service that you configure from the *Telephony Features Permissions* edit page, shown in Figure 10-3. Telephony features permissions are assigned to user groups and define how users can use their telephone features, such as call stack depth, paging, and call forwarding to an external number.

Class of Service
Edit Telephony Features Permissions

[New](#) [Copy](#) [Save](#) [Delete](#) [Reset](#)

[Help](#)

Edit this record

Name:

Max. Call Stack Depth:

Max. Parties in Make Me Conference:

Allow Call Pickup

Allow Trunk-to-Trunk Transfer

Allow Overhead and Group Paging

Allow Make Hunt Group Busy

Allow Extension Reassignment

Allow PSTN Failover

Show Caller ID Name and Number on Monitored Extensions

Allow Customization of IP Phone Buttons and Call Manager Monitor Windows

Show Extensions with Different Prefixes in Directory

Allow Collaboration Features

Allow Recording of Own Calls

Directed Intercom / Paging:

Barge In:

Record Other's Calls:

Monitor Other's Calls:

Allow Call Handling Changes:

Allow External Call Forwarding and Find Me Destinations

Allow External Extension Reassignment

Scope:

Local Only

National Long Distance

National Mobile

International Long Distance

All Calls

Restrictions:

Permissions:

[Refresh this page](#)

Accept: None All Only From: [Search](#)

Accept: None All Only From: [Search](#)

Accept: None All Only From: [Search](#)

Accept: None All Only From: [Search](#)

Current Mode Detailed Settings

(e.g. +1900,+1xxx976)

(e.g. 1/311)

Figure 10-3 Telephony Features Permissions Edit Page

10.2.1.1 Parameters

The parameters on the *Telephony Features Permissions* edit page are as follows:

- **Name:** This lets you specify a descriptive name of the CoS that you are creating or editing.

- **Max. Call Stack Depth:** This lets you specify the maximum number of simultaneous calls that can be “stacked” on a user’s extension. When this number is met, additional inbound calls are routed to the call forward busy destination.
Valid entries are 1 through 16
- **Max. Parties in Make Me Conference:** This lets you specify the maximum number of parties that can be included in any Make Me Conference call made from your site. Select the number of parties from the drop-down. The default number is 3. Valid values are 3 through 6.
- **Allow Call Pickup:** This check box enables call pickup, which lets users pick up any ringing extension (including the night bell) or pick up any parked call.
- **Allow Trunk-to-Trunk Transfer:** This check box enables trunk-to-trunk call transfers. A trunk-to-trunk transfer is defined as follows:
 - An internal party is talking with an external party. The internal party transfers the external party *blindly* or *consultatively* to an external party via the telephone or a ShoreWare Call Manager application.
 - A three-party conference call with one internal party and two external parties when the internal party drops from the call via the telephone or a ShoreWare Call Manager application.

A trunk-to-trunk transfer is not an external party forwarded to an external number by a user’s call handling mode. This is not to be confused with **Allow External Call-forwarding**.

This feature is automatic and by-passes toll-related call permissions. If there is the potential for toll fraud by employees who might abuse this feature, limit this permission to only a few select user groups. For example, you might only give permission to the executive and sales user groups.

If this parameter is enabled, you can manage trunk-to-trunk transfers from the *Call Control Options* edit page.

- **Allow Overhead and Group Paging:** This check box lets users dial any site paging extension and make an announcement using the overhead paging system or group paging.
- **Allow Make Hunt Group Busy:** If granted, this permission allows the user to busy out a hunt group or return it to service by keying *18 on the telephone keypad. In addition, the Quick Look maintenance page can be used to busy out or return a hunt group to service. Calls will be forwarded to the Busy Destination set in the Hunt Groups page.
Should a hunt group be busied out while on a holiday or off-hours schedule, the schedule takes precedence.
- **Allow Extension Reassignment:** This check box lets users reassign their extension to another telephone.
- **Allow PSTN Failover:** This check box indicates that site-to-site calls that fail over proprietary routes will be automatically made over a PSTN number. The PSTN number to use in case of failure must be entered on the User edit page.
- **Show Caller ID Name and Number on Monitored Extensions:** Select this check box if you would like the monitoring party to see information about the incoming calls on the monitored party's line. Leave this check box clear if you do not want the monitoring party to know who the monitored party is talking to. If unchecked,

the incoming call will still be shown with an icon and LED, but not textually in the announcement area.

- **Allow Customization of IP Phone Buttons and Call Manager Monitor Windows:** Select this check box to allow users to configure the programmable buttons on their ShorePhone or BB24 device.
Clear the check box to prevent users from being able to configure custom buttons. This action will, for example, prevent users from being able to configure their phones to monitor the extension of anyone they please and will require that extension monitor, and other features, be set up by a system administrator.
Without the box checked, only the system administrator can configure a user's phone for extension monitoring. Extension Monitor will only monitor those extensions that are monitored on the phones. If there are none, extension Monitor is disabled.
- **Show Extensions with Different Prefixes in Directory:** Select this check box to display extensions that have different prefixes. The On-Net Dialing feature often results in a remote site having a different prefix from a headquarters site. Enabling this check box will cause all extensions (even those at the remote sites) to appear in the directory.
- **Allow Collaboration Features:** Checking this box allows document sharing. See also Chapter 11, starting on page 245.
- **Allow Recording of Own Calls:** Checking this box grants permission to record one's own calls. This will also be impacted by the Enable Monitor/Record Warning Tone parameter in Chapter 9, starting on page 133.
- **Directed Intercom/Paging:** The *Allow initiation* check box and *Accept* radio buttons can be used to configure the intercom/paging calls function.
See Section 15.1.5.1 on page 302 for details on configuring this option.
- **Barge In:** Barge In permits one party to join an existing call as a fully conferenced participant. When barge in is initiated, a brief intrusion tone is played to the other participants and (if present) the monitor/record warning tone is discontinued. Checking *Allow initiation* allows users with this CoS to barge in on users with *Accept* Barge In enabled.
See Section 15.1.5.1 on page 302 for details on configuring this option.
- **Record Other's Calls:** Select this check box to allow a user to record the calls of another user (e.g. allowing a supervisor to record the calls of an agent in a call center environment).
The "Selectable Mailboxes" feature allows the automatic placement of recorded calls into mailboxes other than that of the client who initiated the call recording. See "Record Call" and "Record Extension" in Table 10-1 on page 207 for details.
A monitor/record warning tone is played. To make recordings silent, disable the warning tone. See Section 9.8.2.1 on page 154 for details.
See Section 15.1.5.1 on page 302 for details on configuring this option.
- **Monitor Other's Calls:** These check boxes allow the user to monitor calls. The monitor/record warning tone is played, if enabled in Call Control.
See Section 15.1.5.1 on page 302 for details on configuring this option.
- **Allow Call Handling Changes:** These check boxes allows users to make changes to their call handling settings.

- **Current Mode**—Checking this box allows users to change their current call handling mode from ShorePhone IP phones and Personal Call Manager.
- **Detailed Settings**—Checking this box allows users to change all call handling settings, such as call forward destinations, from Personal Call Manager.
- **Allow External Call-forwarding and Find Me Destinations:** This check box lets users forward incoming calls to an external number, as restricted by the *Scope*, *Restrictions*, and *Exceptions* parameters.

This must not be confused with *Allow Call Transfer Trunk to Trunk*.

- **Allow External Extension Reassignment:** This check box allows users in this CoS to assign their extension to a PSTN phone for use with the Office Anywhere feature. By selecting this check box, the user will be able to treat his cell phone or home phone as an extension on the ShoreTel 6.1 system.
- **Scope:** Scope allows you to set a general permission level. Levels are presented from the most restrictive to the most permissive. The Restrictions and Permissions listed are applied in addition to the general scope setting.
 - **Local Only** allows forwarding or extension reassignment only to local or additional local area codes, as defined on the *Site* edit page.
 - **National Long Distance** also allows forwarding or extension reassignment to long-distance numbers within the country, as defined on the *Site* edit page.
 - **National Mobile** allows forwarding or extension reassignment to mobile numbers. Some countries use “caller pays” mobile calling, so do not select this radio but if you do not want to incur the associated costs of calls being sent to a mobile phone.
 - **International Long Distance** also allows forwarding or extension reassignment to international numbers, as defined on the *Site* edit page.
 - **All Calls** allows forwarding or extension reassignment to any number, including 900, Operator Assisted, and Carrier Select numbers. This is the default.
- **Restrictions:** The Restrictions and Permissions listed are applied in addition to the Scope setting. The comma-separated restriction expressions are limited to a total of 50 characters.

Follow these guidelines for entering restrictions:

- Numbers must be entered in canonical format, including the international designation “+” and country code. For example, to restrict forwarding to the 408 area code in the U.S., use +1408.
- Non-routable calls (311, 411, etc.) for a country must be designated by the country code plus the “/” character. For example, to restrict forwarding to 311 in the U.S., use 1/311.
- Each field can contain multiple entries as long as they are separated by commas or semicolons.
- Each entry must consist of numbers only.
- Access codes, such as 9, must not be included.
- The wildcard of “x” can be used.

When a call is both restricted and permitted, it is permitted. For example, restricting +1 408 and permitting +1 408 331 restricts all calls to the 408 area code except those to 408 331-xxxx.

- **Permissions:** The Restrictions and Permissions listed are applied in addition to the Scope setting. The comma-separated restriction expressions are limited to a total of 50 characters.

Follow these guidelines for entering permissions:

- Numbers must be entered in canonical format including the international designation “+” and country code. For example, to permit forwarding to the 408331 area code and prefix in the U.S., use +1408331.
- Non-routable calls (311, 411, etc.) for a country must be designated by the country code plus the “/” character. For example, to permit forwarding to 311 in the U.S., use 1/311.
- Each field can contain multiple entries as long as they are separated by commas or semicolons.
- Each entry must consist of numbers only.
- Access codes, such as 9, must not be included.
- The wildcard of “x” can be used.

When a call is both restricted and permitted, it is permitted. For example, restricting +1 408 and permitting +1 408 331 restricts all calls to the 408 area code except those to 408 331-xxxx.

10.2.2 Whisper Page

The Whisper Page feature allows an operator to break into an active call in order to speak with an internal user. This occurs without the remote caller hearing the interruption and without the operator hearing the remote caller.

Here's a real-world example: You are in the office on a call with a client when another client arrives in the lobby to meet you for an appointment. The administrative assistant knows that you are on a call, but she uses the Whisper Page feature to interrupt the call to let you know someone is waiting for you in the lobby. You hear the voice of your administrative assistant and the client at the same time, but neither of them can hear one another.

Implementation details:

- The Whisper Page feature can be invoked from:
 - Personal Call Manager (PCM)
 - Any phone (analog or IP) by pressing the code *19
 - One of the ShorePhone soft keys
- While on a Whisper Page call, the internal user can mute the audio channel to the original caller. This allows him to respond to the operator without the original caller hearing him. This can be accomplished from:
 - One of the ShorePhone soft keys - not the standard mute button
 - PCM, if you do not have an IP phone
- Both the operator and the internal user hear a tone when the Whisper Page call is connected. The tone is the same as the one used for the intercom feature.

- To receive a Whisper Page call, the internal user must be on the handset of a multiline ShorePhone. If a Whisper Page call is sent to any other phone (e.g., IP100, SoftPhone, or analog phone) the call will be treated as an intercom call.
- If a Whisper Page call is sent to a phone that is not on an active call, the feature behaves the same as an intercom call.
- Permissions to use this feature are the same as for intercom, and the feature is enabled or disabled (via Director) along with the intercom feature.
- The Whisper Page feature will not work if the internal party is on a 3-way conference call.
- No call control operations may be performed on the Whisper Page call, except to hang up. For example, the Whisper Page call cannot be put on hold, transferred, parked, etc.

Configuring Whisper Page

To configure the Whisper Page feature, do the following:

- 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 Users link and then the Class of Service link. The Class of Service window appears.
- Step 4** Select the name of the one of the profiles in the Telephony Features Permissions section by clicking on it. (You can either create a new profile or modify one of the existing profiles.)
- Step 5** The Class of Service window appears, as shown below.
There is no separate check box specifically for the Whisper Page feature. The functionality has been coupled with the intercom functionality.
- Step 6** In the Directed Intercom/Paging field, enable the Allow Initiation check box to allow this user to place a Whisper Page call. (This user would most likely be an operator or administrator.) You can also enable the Accept check box if the user is likely to be on the receiving end of a Whisper call.
- Step 7** Click the *Save* button to store your changes.

The topic of applying a Class of Service to a User Group is covered in the System Administration Guide. However, as a reminder, keep in mind that once you have enabled the Whisper Page feature in a Class of Service profile, you must make sure that this Class of Service profile is applied to the appropriate User Group to which you target user belongs. In other words, the user that will be using the Whisper Page feature must belong to a User Group that has the COS - Telephony value set to the name of the profile that you just finished modifying.

10.2.3 Call Permissions

This section describes the classes of service that configured in the *Call Permissions* edit page, shown in Figure 10-5. Call permissions are assigned to user groups, providing users with specific telephone calling permissions.

Class of Service
Edit Telephony Features Permissions

[New](#) [Copy](#) [Save](#) [Delete](#) [Reset](#) [Help](#)

[Edit this record](#) [Refresh this page](#)

Name:

Max. Call Stack Depth:

Max. Parties in Make Me Conference:

Allow Call Pickup

Allow Trunk-to-Trunk Transfer

Allow Overhead and Group Paging

Allow Make Hunt Group Busy

Allow Extension Reassignment

Allow PSTN Failover

Allow Collaboration Features

Allow Recording of Own Calls

Directed Intercom / Paging: Allow Initiation Accept

Barge In: Allow Initiation Accept

Record Other's Calls: Allow Initiation Accept

Monitor Other's Calls: Allow Initiation Accept

Allow Call Handling Changes: Current Mode Detailed Settings

Allow External Call Forwarding and Find Me Destinations

Scope: Local Only
 National Long Distance
 International Long Distance
 All Calls

Restrictions: (e.g. +1900,+1xxx976)

Permissions: (e.g. 1/311)

Figure 10-4 Enable Whisper Page

Class of Service
Edit Call Permissions

[New](#) [Copy](#) [Save](#) [Delete](#) [Reset](#) [Help](#)

[Edit this record](#) [Refresh this page](#)

Name:

Scope: Internal Only
 Local Only
 National Long Distance
 International Long Distance
 All Calls

Restrictions: (e.g. +1900,+1xxx976)

Permissions: (e.g. 1/311)

Figure 10-5 Call Permissions Edit Page

10.2.3.1 Parameters

The parameters on the *Call Permissions* edit page are as follows:

- **Name:** This is a descriptive name of the Class of Service record that you are adding or editing.
- **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.
- **National Long Distance** also allows calls to long-distance numbers within the country, as defined on the *Site* edit page.
- **International Long Distance** also allows calls to international numbers, as defined on the *Site* edit page.
- **All Calls** allows calls to any number, including 900, Operator Assisted, and Carrier Select numbers, as well as use of Vertical Service Codes. This is the default.
- **Restrictions:** The Restrictions and Permissions listed are applied in addition to the Scope setting. The comma-separated restriction expressions are limited to a total of 50 characters.

Follow these guidelines for entering restrictions:

- 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.
- 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.
- The wildcard of “x” can be used.

When a call is both restricted and permitted, it is permitted. For example, restricting +1 408 and permitting +1 408 331 restricts all calls to the 408 area code except those to 408 331-xxxx.

- **Permissions:** The Restrictions and Permissions listed are applied in addition to the Scope setting. The comma-separated restriction expressions are limited to a total of 250 characters.

Use the following guidelines for entering permissions:

- In general, numbers must be entered in canonical format including the international designation “+” and country code. For example, to permit calls to the 408331 area code and prefix in the U.S., use +1408331.
- Non-routable calls (311, 411, etc.) for a country must be designated by the country code plus the “/” character. For example, to permit 311 in the U.S., use 1/311.
- Each field can contain multiple entries as long as they are separated by commas or semicolons.
- Each entry must consist of numbers only.
- Access codes, such as 9, must not be included.
- The wildcard of “x” can be used.

When a call is both restricted and permitted, it is permitted. For example, restricting +1 408 and permitting +1 408 331 restricts all calls to the 408 area code except those to 408 331-xxxx.

10.2.4 Voice Mail Permissions

This section describes the classes of service that you configure from the *Voice Mail Permissions* edit page, shown in Figure 10-6. Voice mail permissions are assigned to user groups, providing users with specific usage of the ShoreTel 6.1 voice mail system.

Figure 10-6 Voice Mail Permissions Edit Page

10.2.4.1 Parameters

The parameters on the *Voice Mail Permissions* edit page are as follows:

- **Name:** This is a descriptive name of the CoS record that you are creating or editing.
- **Incoming Message Length:** This is the maximum length of an incoming voice mail message.
Valid values are from 1 to 3600 seconds. The default is 300 seconds.
- **Incoming Max. Messages:** This is the maximum number of messages that can be queued in a mailbox, including new, saved, and deleted messages.
Valid values are from 1 to 500 messages.
- **Outgoing Message Length:** This is the maximum message length that a user can record before sending a message to another extension. This parameter controls both the composed message and the greeting.
Do not confuse this with the user's personal voice mail greeting.
Valid values are from 1 to 3600 seconds.
- **Delete Saved / Unheard Messages after (n) days:** Select this check box to have the system automatically delete saved or unheard voice mail messages that are older than *n* days.

Valid values are from 30 to 2000 days. The default value for the time limit is 0 days (meaning the feature is disabled).

If enabled, users who have messages older than the expiry time limit will receive multiple warnings in the form of email and/or voice mail messages indicating that their voice mail messages (that exceed the time limit) will be deleted. The first warning is sent two weeks before the expiration deadline, and a second message is sent one week prior to deadline.

After the time limit has passed, the user will receive a message similar to the following: “*Your mailbox was cleaned automatically; n messages were deleted.*”

This message can be emailed to the user if an address has been configured in *Director > Edit User > General* tab.

- **Delete Heard Messages after (n) days:** Select this check box to have the system automatically delete heard voice mail messages after *n* days.

Valid values are from 30 to 2000 days. The default value for the time limit is 0 days (meaning the feature is disabled).

As with Saved/Unheard messages, the user will receive several warnings before messages are deleted (see above for details).

- **Lifespan of Voicemail Password:** Select this check box to have the system periodically require users to change their TUI passwords. It is recommended to enable this feature as requiring periodic password changes increases the security of the system.

The password change applies to the following Dialed Number types:

- User extensions
- Workgroup extensions
- Route point extensions
- External user extension

Valid values are from 30 to 365 days. The default value for the time limit is 90 days.

- **Days in Advance of Password Expiration Before Warning:** When the Lifespan of Voicemail Password is selected, this option to warn users that their password is about to expire also becomes active. You can select the check box to enable the feature, thus allowing users to be notified prior to the expiration of their passwords, and giving them time to proactively change their passwords days or weeks in advance of the actual expiration date.

Simply enable the feature by selecting the check box and then enter the number of days ((prior to the password expiration), at which point group members will begin receiving warning messages. If the check box is not selected or if 0 is placed in the data entry field, password expiration warning messages are not delivered to the group member.

Valid values are from 1 to 30 days. The default value for the time limit is 7 days.

- **Allow Access to Broadcast Distribution List:** This gives users access to the company-wide distribution list. A user with this permission is able to broadcast voice mail messages to all users throughout the company.
- **Allow Access to System Distribution Lists:** This gives users access to system distribution lists.

- **Allow Message Notification:** This enables message notification. It might be further qualified by *Allow Message Notification to External Number*.

This is enabled as a system default.

- **Allow Message Notification to External Number:** This permits message notification to an external number. It cannot be set unless *Allow Message Notification* is enabled.

This is enabled as a system default.

10.3 User Groups

Configuring user groups is the second step in configuring users. The *User Groups* page, shown in Figure 10-7, lets you create user groups and edit existing user group configurations. To access the *User Groups* list page, click *User Groups* from the navigation frame.

User Groups							Help
Add new							
Permissions							
Name	Telephony Features	Call	Voice Mail	SMDI	Account Codes	DID As CESID	
Account Code Service	Minimally Featured	No Restrictions	No Mail Box	None	Disabled	Yes	
Anonymous Telephones	Minimally Featured	Internal Only	No Mail Box	None	Disabled	Yes	
Executives	Fully Featured	No Restrictions	Large Mail Box	None	Disabled	Yes	
House Telephones	Minimally Featured	Internal Only	No Mail Box	None	Disabled	Yes	
IP Telephones	Minimally Featured	Internal Only	No Mail Box	None	Disabled	Yes	
Managers	Partially Featured	No Restrictions	Medium Mail Box	None	Disabled	Yes	
Staff	Partially Featured	Long Distance Calling	Medium Mail Box	None	Disabled	Yes	
Voice Mail Notification	Voice Mail Featured	No Restrictions	No Mail Box	None	Disabled	Yes	

Figure 10-7 User Groups List Page

If you have not created classes of service for your user groups, refer to Section 10.2 on page 174 and configure them.

10.3.1 User Group List Page Column Definitions

The columns in the *User Groups* table are as follows:

- **Name:** This is the name of the user group.
- **Telephony Features:** This is the telephony features permissions CoS associated with the user group.
- **Call:** This is the call permissions CoS associated with the user group.
- **Voice Mail:** This is the voice mail CoS associated with the user group.
- **SMDI:** This is the Simplified Message Desk Interface mode set for the user group. Also see Section 4.3 on page 47.
- **Account Codes:** This is the account code collection mode set for the user group. Also see Section 9.2 on page 133.
- **DID As CESID:** Indicates whether a DID number should be sent as the Caller's Emergency Service ID number for a 911 emergency call.

When you click *Add new* or click the name of an existing user group from the *User Groups* page, the *User Group* edit page, shown in Figure 10-8, appears.

The screenshot shows the 'User Groups' edit page for a user group named 'Account Code Service'. The page has a title bar with 'User Groups' and 'Edit User Group', and buttons for 'Save' and 'Reset'. A 'Help' link is in the top right. Below the title bar is a 'Refresh this page' link. The main form contains several fields and checkboxes:

- Name:** Account Code Service
- COS - Telephony:** Minimally Featured (with a 'Go to this Class of Service' link)
- COS - Call Permissions:** No Restrictions (with a 'Go to this Class of Service' link)
- COS - Voice Mail:** No Mail Box (with a 'Go to this Class of Service' link)
- Simplified Message Desk Interface Mode:** None
- Account Code Collection:** Disabled
- Show Call Manager users a list of account codes when dialing.
- Send Caller ID as Caller's Emergency Service Identification (CESID).
- Send DID as Caller's Emergency Service Identification (CESID).
- Outgoing Trunk Groups (Access Code):** A list of five items, all checked:
 - Analog Loop Start (9)
 - Digital Loop Start (9)
 - Digital Wink Start (9)
 - PRJ (9)
 - Primary (9)
 - San Jose (9)

A note at the bottom states: 'Note: This user group is for Account Code Service only.'

Figure 10-8 User Group Edit Page

10.3.2 User Group Parameters

The parameters on the *User Group* edit page are as follows:

- **Name:** This is a descriptive name of the group you are adding or editing.
- **COS – Telephony:** This is the telephony features permissions CoS associated with the user group.
- **COS – Call Permissions:** This is the call permissions CoS associated with the user group.
- **COS – Voice Mail:** This is the voice mail permissions CoS record associated with the user group.
- **Simplified Message Desk Interface Mode:** Select from *None*, *External Voice Mail*, or *ShoreTel Voice Mail*. Select *External Voice Mail* if an application server is functioning as a PBX for a legacy voice mail system. Select *ShoreTel Voice Mail* if an application server is functioning as a voice mail server for a legacy PBX. Also see Section 4.3 on page 47.
- **Account Code Collection:** Select from *Disabled*, *Optional*, or *Forced* account code collection for the selected user group.
 - *Disabled*—Account collection is not active for this group.

- **Optional**—Users are prompted to enter an account code. If no account code is entered, the call is completed without account code records.
- **Forced**—Users must enter an account code for all calls outside the bounds of the call permissions set for the user.

For more information about account codes, see Section 9.2 on page 133.

- **Show Call Manager users a list of account codes when dialing:** Enabling this option allows Call Manager users to select an account code from the complete list of account codes when prompted for an account code. Disable this feature if you want to restrict the user's knowledge of account codes.
- **Send Caller ID as Caller's Emergency Service Identification (CESID):** The caller ID configured on the User's page is the telephone number sent to the service provider when a user dials an emergency services number (e.g., 911 in the U.S.). Default setting is checked on. If this option is not selected, the outbound caller ID will be either the user's DID or the site's CESID.

For more information on emergency and 911 calls, see Appendix B, starting on page 399, for more information.

- **Send DID as Caller's Emergency Service Identification (CESID):** This is the telephone number sent to the service provider when a user dials an emergency services number (e.g., 911 in the U.S.) from their home phone. If this option is not selected and Send Caller ID as Caller's Emergency Service Identification (CESID) is also not selected, the outbound caller ID will be the site's CESID.

For more information on emergency and 911 calls, see Appendix B, starting on page 399, for more information.

- **Outgoing Trunk Groups (Access Code):** These are the trunk groups to which this user group has access for outgoing calls.

For more information about trunk groups, see Chapter 7, starting on page 95.

You can assign multiple trunk groups with the same access code to the user group.

10.4 Individual Users

Configuring individual users is the final step in setting up user parameters. From the *Individual Users* list page, shown in Figure 10-9, you can invoke the *Edit User* page by selecting a site and clicking *Go*. The resulting *Edit User* page lets you specify the user's general information, configure personal options, and edit distribution lists.

If you have not configured user groups for your users, refer to the Section 10.3 on page 185 and configure them first.

If there are more users than can be displayed on one page, you can use forward and backward arrows to page through the names and specify the number of records to display per page.

10.4.1 Individual Users List Page Column Headings

The columns in the *Individual Users* list page are defined as follows:

- **First Name:** The first name of the user.
- **Last Name:** The last name of the user.
- **Site:** The site associated with the user.

First Name	Last Name	Site	User Group	Client Type	Extension	Mailbox	Switch	Port	Status
Adam	Ant	Headquarters	Executives	Operator	4509	4509	Doc-24	ShoreDoc IP Phone	Home
Betty	Boop	Headquarters	Executives	Operator	4510	4510	Doc-24	10	Home
Carl	Castle	Headquarters	Executives	Operator	4511	4511	Doc-24	11	Home
Donald	Duck	Headquarters	Executives	Operator	4512	4512	ShoreDoc-HQ	Any IP Phone	Home
Ernie	Els	Headquarters	Executives	Personal	4513	4513	Doc-24	13	Home
Fred	Flintstone	Headquarters	Executives	Personal	4514	4514	Doc-24	14	Home
Greta	Garbo	Headquarters	Executives	Personal	4515	4515	Doc-24	15	Home
Howard	Hughes	Headquarters	Executives	Personal	4516	4516	Doc-24	16	Home
Johnathan	Seagull	Headquarters	Executives	Personal	4805	4805	Doc-24	1	Home
Tim	Toolman	Headquarters	Executives	Personal	4904	4904	Bedrock	10	Home
Ursula	Urp	Headquarters	Executives	Personal	4907	4907	Bedrock	13	Home
Vince	Vinceble	Headquarters	Executives	Personal	4906	4906	Bedrock	12	Home
Wanda	Wenderly	Headquarters	Executives	Personal	4908	4908	Bedrock	14	Home
Wendy	Williams	Headquarters	Executives	Personal	4905	4905	Bedrock	11	Home
Yourst	Yumbal	Headquarters	Executives	Personal	4903	4903	Bedrock	9	Home

Figure 10-9 Users List Page

- **User Group:** The user group that is associated with the user.
- **Client Type:** The client type that is assigned to the user.
- **Extension:** The user's extension.
- **Mailbox:** The user's voice mailbox.
- **Switch:** The switch associated with the user.
- **Port:** The port associated with the user or the MAC address of an IP phone.
- **Status:** This shows the user's telephone port status. *Home* indicates that the user is at his or her home telephone port; *Assigned* indicates otherwise.

To add or edit a user's profile from the *Edit User* page, fill out the four tabs—*General* information, *Personal Options* for user options, *Distribution Lists*, and *Workgroups*—as described in the following sections. For information on importing user information from a spreadsheet, see Section 10.4.3 on page 189.

10.4.2 Private Numbers

Users can have private numbers that are not listed in the System Directory or in Call Manager Quick Dialer and thus have Caller ID information suppressed. This is enabled through the check box on the *Edit User* page called Make Number Private. When checked, the user's extension becomes a Private Number.

The following conditions apply to private numbers:

- The user's Private Number extension no longer appears in the QuickDialer for dial-by-name operations or in the ShoreTel Directory Viewer.
- Calls placed by a Private Number user show the caller's name but not their number to the dialed party for internal calls. This applies to analog phones, IP phones, and associated instances of the Call Manager. The ring style is a double-ring, indicating an internal call.
- Calls placed from a private number to an external party do not deliver a Direct-Inward-Dial (DID) number as Caller ID when PRI trunks are used for the outbound

call. The proper CESID (Caller's Emergency Service ID) will only be delivered for 911 calls.

- Calls placed from a private number to an off-system extension on PRI trunks with NI2 signaling deliver calling name information but not calling number information.
- Routing slips and the Call Manager and History viewer show the Private Number user's name but not their extension number.
- The Private Number users are listed with name and number in the Extension Monitor extension selection dialog box.
- The user can be dialed directly via the telephone or the Call Manager if their extension is known.
- Contacts imported from Microsoft Outlook or Exchange that reference a Private Number user's extension are not blocked and are fully visible in the Call Manager Quick Dialer.
- CDR database records show both number and name for Private Number users. The Caller-ID Flags field will indicate that only the name is valid however.
- CDR legacy log files show the number of Private Number user calls that are inbound or outbound calls.
- The ShoreWare Director shows number information for Private Number users as with other users, as for example on the User List page.

10.4.3 User Import Tool

The database import utility allows a system administrator to make changes across all users in a system with a single action. This new feature improves the ease with which a system administrator can modify information for large groups of users.

The system administrator creates a spreadsheet using a common application, such as Microsoft Excel. He or she modifies the user data in the spreadsheet in “free form” fashion, instead of having to modify each user's information, one at a time, from within Director. For someone who is skilled at using spreadsheet software, the time savings can be enormous.

Once the user information has been manipulated within the spreadsheet, it can be exported to a CSV format and this CSV file can then be imported into the ShoreTel 6.1 system.

Details:

- The import tool supports modify/delete/insert operations, allowing a system administrator to add users, delete users, or modify the account of an existing user.
- Using the import tool does not require scheduled downtime. However, when importing large files that contain many rows of information, performance may be affected as the database is frequently queried. So depending on the size of the imported file and the type of information that is being added or modified, it may be recommended to perform the import during off hours.
- The ShoreTel 6.1 system does not currently support the ability to export user data into a CSV file.
- The Import utility does not allow the administrator to specify the Users' Home Port. The *Home Port* parameter is automatically set to SoftSwitch within each User record added or modified by an import.

10.4.3.1 Additional CSV Details:

The following bullets represent a set of field headers that must appear as the first row in the CSV file that will be imported. The field order can be random, but all fields are mandatory, case sensitive, and must have data:

- **Extension** (-1 for auto generated)
- **FirstName** (String)
- **LastName** (String)
- **UserLicenseType** (Code)
 - Extension-Mailbox
 - Extension-Only
 - Mailbox-Only
- **CallerID** (String)
 - A full canonical number (such as +1 (408) 331-3300)
- **UserGroup** (Code. Must be one of the existing User groups configured on the system.)
- **Site** (String Name - Must be one of the existing Site Name)
- **Language** (Code. Must be one of the following:)
 - English (US)
 - Spanish
 - German
 - English (UK)
 - French
- **VMServer** (String Name - Must be one of the existing Voice-mail Server Name)
- **CallStackSize** (Number)
- **AcceptBroadcasts** (Boolean)
- **MakeNumberPrivate** (Boolean)
- **DialByName** (Boolean)
- **AllowSoftPhone** (Boolean)
- **ClientType** (Must be one of the following:)
 - Professional
 - Personal
 - Workgroup Supervisor
 - Workgroup Agent
 - Operator
 - Contact Center Supervisor
 - Contact Center Agent
- **GuiLoginName** (String)
- **GUIPassword** (String)
- **TUIPassword** (String)
- **EmailDomain** (String)

- **ConferenceServer** (String Name - Must be the existing Conference Bridge Name or "None")
- **ConferenceServerUserID** (String - User ID for the conference bridge)
- **ConferenceServerPassword** (String)
- **RingType** (Code. Must be one of the following:)
 - Standard
 - Ring 2
 - Ring 3
 - Ring 4
- **CallWaitingToneEnabled** (Boolean)
- **FaxSupport** (Must be one of the following)
 - User - No Redirect
 - User - Redirect
 - Fax Server
 - Fax Machine
- **HeadsetAudioPath** (Boolean)
- **HeadsetMode** (Boolean)
- **PSTNFailOverType**
 - None
 - External
 - DID
- **PSTNFailOverNumber**
 - A full canonical number (such as +1 (408) 331-3300)

Formatting in the Excel spreadsheet is removed when the sample file is saved to CSV format. In some cases, this loss of formatting may result in the sample file containing a number that is in an invalid format, instead of in the required canonical format.

 - Used in case PSTNFailOverType is set to 2 (External)
- **DIDRange**
 - A full canonical number base of range (such as +1 (408) 331-3300)
- **DIDNumber**
 - A full canonical number (such as +1 (408) 331-3300)
- **VoiceMailboxForRecordedCalls**
 - An existing valid extension (or can be left blank)

Additional considerations:

- String names fields must exist in the system. For example, if a new user is to be created in site New York. "New York" must exist as a site name. String names are case sensitive.
- For fields that are code in nature, you must specify the displayable description as it appears in Director. Data validation will translate the displayable value to the appropriate code. Note that the descriptions are case sensitive.

- Boolean fields can be true/false or 1/0.
- All fields are mandatory when adding new users.
- When updating an existing user, fields left blank will not change existing values.

Example: The following shows a sample of the spreadsheet values before the Director import.

1	Extension	FirstName	LastName	GuiLoginN	GUIPassw	TUIPassw	UserLicen	CallerID	UserGroup	Site	Language	VMServer	CallStack	AcceptBro
2	233	Jane	Doe	Changeme	1234	3456	Mailbox-Only		Executives	Headquart	English(US	Headquart	2	TRUE

Figure 10-10 Spreadsheet values before Director import

The following procedure describes the process of importing data from a CSV file into the user database. This procedure assumes that you have already successfully exported the data from the spreadsheet into a CSV file, and stored that file in the proper location.

- Step 1** Verify that the CSV file that will be imported is located on the HQ server. It should be located in the following directory:
C:\Program Files\Shoreline Communications \ShoreWare Server
- Step 2** Open the command prompt window in the directory shown above and run the following command:
`Dbimport -log DbLog.log -err DbErr.err <CSV-file>`
- `-err` is the flag to create an error log file.
 - `error.log` is the file where error messages will be stored.
 - `CSV-file` is the name of the file.

The import adds any new users and modifies any existing users in the user database as indicated in the CSV file.

When a user is updated, he or she is assigned to “Any IP Phone” on the headquarters server.

10.4.4 General Information

General information about new and existing users is provided under the *General* tab on the *Edit User* page. The following three subsections describe parameters on the General panel.

10.4.4.1 Upper-Section Parameters

The parameters in the upper portion of the *Edit User: General* panel, are shown in Figure 10-11.

The following are description of parameters located in the upper section of the *Edit User: General* panel.

- **First Name:** Specifies the first name of a user, fax machine, conference room, or “virtual user.”
- **Last Name:** This is the last name of the user.
- **Number:** This is the work number for the user. The number defaults based upon a cookie that is locally stored since the last time you saved a user.
- **License Type:** Set the user’s license type from the following values: Extension and Mailbox, Extension-Only, and Mailbox-Only.

The screenshot shows the 'Edit User' interface in the ShoreTel administration console. The 'General' tab is selected, and the 'Upper section' of the form is visible. The form contains the following fields and values:

- Buttons: New, Copy, Save, Delete, Reset, Help
- Tabs: General (selected), Personal Options, Distribution Lists, Workgroups, Refresh this page
- First Name: Little
- Last Name: Bear
- Number: 65701
- License Type: Extension and Mailbox
- Caller ID: (empty) (e.g. +1 (408) 331-3300)
- DID: DID: +14085 (DID Range: +14085413936 - 413940)
- PSTN Failover: None
- User Group: Executives (Go to this User Group)
- Site: HQ
- Language: English(US)
- Home Port:
 - IP Phones (Any IP Phone)
 - Ports (HQ 50V - 11)
 - SoftSwitch (SoftSwitch)
- Current Port: SoftSwitch (Go Home)
- Jack #: (empty)

© 1998-2008 ShoreTel, Inc. All rights reserved.

Figure 10-11 Edit User: General panel – Upper section

- **Extension and Mailbox**—The user will have both a phone extension and an internal ShoreTel mailbox for voice mail.
- **Extension-Only**—The user will have an extension but no mailbox. Use Extension-Only if, for example, the user has a mailbox for voice mail on a legacy switch or PBX. Selecting Extension-Only results in the following:
 - * The voice mail server drop-down menu and associated check boxes are disabled.
 - * User groups with SMDI ShoreTel voice mail are not available.
- **Mailbox-Only**—The user will have a mailbox on the ShoreTel switch but will not have a phone extension. Selecting Mailbox-Only results in the following:
 - * User groups with SMDI External voice mail are not available
 - * Ports cannot be assigned to Mailbox-Only users.
 - * Office Anywhere is not available to Mailbox-Only users, regardless of CoS settings.

ShoreTel 6.1 capacity is licensed by user license type. Make sure you have licenses for all users and required types.

- **Caller ID:** This number is used for caller ID on outbound calls. A caller ID entered here will take precedence over the user's DID and the site's CESID number (both for normal outbound and 911 calls). If no number is entered, the user's DID or the site's CESID will be used for outbound caller ID.

This feature is available only on outbound calls using a T1 PRI trunk.

- **DID:** This provides the user with a Direct Inward Dial (DID) number. The number defaults based upon a cookie that is locally stored since the last time you saved a

user. This requires that DID services are properly configured against the desired trunk group.

This is enabled by default if a DID trunk group is configured.

- **PSTN Failover:** Select from None, DID, or a specific phone number. When a specific phone number is to be entered, a text box opens. This field supplies the PSTN number to be dialed to complete a failed site-to-site call. When DID is selected and a call fails, the call is routed via the PSTN network to a local PSTN DID number for the user. When an external number is supplied and a call fails to go directly to its intended extension, the call is routed to the number indicated, for example, an auto-attendant.

The user must have a Class of Service that has “Allow PSTN Failover” enabled. See Section 10.2 on page 174.

If there is no available bandwidth or if a WAN is down for a site to site call and if the call destination has no PSTN failover, the call is directed to voice mail.

- **User Group:** This displays all of your user groups. Associate one group with the user. The default is *Executives*. Clicking *Go to this User Group* invokes the associated user group page.

If you have not configured your site’s user groups, refer to Section 10.3 on page 185 and configure them before proceeding further.

- **Site:** This is the site for the user. This filters the list of switch ports that you must select from as well as provide a different default DID number if available.
- **Language:** Select the language this user will hear for voice mail prompts from the drop-down list of available languages.
- **Home Port:** This displays a list of available switch ports and IP phones. Select the switch port or IP phone to be associated with this user.

If you are assigning an analog port and do not make a selection, ShoreTel 6.1 selects the next available port.

If you select *IP Phone*, the drop-down list displays *Any IP Phone* as the default. For information about the Any IP Phone feature, see Appendix A, starting on page 391.

If you select *Port*, the drop-down list displays ports available for phone use.

To create a user without a port (a virtual user), select *SoftSwitch* as the home port.

Assigning users to an analog port or SoftSwitch for their home port can cause the loss of phone service if the user selects the Go Home option in Personal Call Manager. ShoreTel recommends that Office Anywhere users be assigned Any IP phone as their home port.

- **Current Port:** This field indicates the user’s current switch port. This shows you which switch port the user has assigned himself or herself. You can change the current port setting to the home port by clicking *Go Home*.

If you click *Go Home*, the system forces the user back to his or her home telephone. This option is useful when a temporary user is no longer using that phone.

- **Jack #:** This is the patch-panel jack number associated with the user’s switch port.

10.4.4.2 Mid-Section Parameters

The parameters in the middle portion of the *Edit User: General* panel, are shown in Figure 10-12.

Users
Edit User

[New](#) [Copy](#) [Save](#) [Delete](#) [Reset](#) [Help](#)

▼ **General** ▶ Personal Options ▶ Distribution Lists ▶ Workgroups [Refresh this page](#)

First Name:
 Last Name:
 Number:

Mailbox on Server: [Escalation Profiles and Other Mailbox Options](#)

Accept Broadcast Messages
 Include in System Dial By Name Directory
 Make Number Private

Fax Support:

Client Type: Enable Contact Center Integration

Allow Video Calls:

Allow Telephony Presence
 Allow Use of Soft Phone
 Allow Phone API
 Allow Mobile Call Manager
 Delayed Ringdown

• Extension:
 • External Number: (e.g. 9+1 (408) 331-3300)

Ringdown Delay: sec

© 1998-2008 ShoreTel, Inc. All rights reserved.

Figure 10-12 Edit User: General panel – Middle section

The following are description of parameters located in the middle section of the *Edit User: General* panel.

- **Mailbox on Server:** This enables the user's voice mailbox. The drop-down list allows you to select the server to host this mailbox. This is enabled by default.
When you change a user's server, the user must exit and re-launch the client to establish a connection to the new server.
- **Accept Broadcast Messages:** When enabled, this lets individual users receive broadcast messages. Disable this feature for users who do not want to receive broadcast messages.
This is enabled by default.
- **Include in System Dial By Name Directory:** When enabled, this causes the user's name to be included in the auto-attendant's dial-by-name directory.
This is enabled by default.
- **Make Number Private:** Checking this check box removes this number from the system directory and call handling destination lists.
- **Fax Support:** If a port is connected to a fax machine and is in an environment where SIP PSTN gateways are used, this option can be configured such that the switch will freeze the jitter buffer and disable the echo canceller at the beginning of the call to ensure that faxes are clearly and reliably transmitted. (While jitter buffer and echo canceller enhance the quality of voice calls, they impede reliable fax transmission.) Options are:

- *User - No Redirect* (Extension is connected to a user; do not redirect inbound fax calls)
- *User - Redirect* (Extension is connected to a user; redirect inbound fax calls to site fax redirect extension)
- *Fax Server* (Extension is connected to a fax server; do not redirect inbound fax calls but pulse DTMF digits)
- *Fax Machine* (Extension is connected to a fax machine; do not redirect inbound fax calls)
- **Client Type:** The client types enable assignment of various levels of permissions for users using the ShoreTel Call Manager application. Select from:
 - *Personal* (default)—This client is available to all users and delivers desktop call control, visual voice mail, call history, and directory services, as well as options to control call handling and message notification. Since this is the default client configuration, it does not require a special license.
 - *Professional* – This client provides access to all functions available through Personal Call Manager plus Instant Messaging, Presence, Contact Viewer, and SoftPhone. A video license is also provided to users that have rights to Professional Call Manager, providing access to VGA video.
 - *Workgroup Agent*—This level is typically assigned to members of a workgroup and provides access to all features available through Professional Call Manager, except video, plus workgroup features, including login, logout, wrap-up, queue monitor, and shared workgroup mailbox.
 - *Workgroup Supervisor*—This level is typically assigned to the supervisor of a workgroup and provides access to the agent monitor, in addition to all features available to Workgroup Agents.
 - *Operator*—This client level is typically assigned to operators, secretaries, and executive assistants. In addition to all features available in Workgroup Supervisor, it provides access to XGA Video and detailed information about destination extensions, including access to an extension monitor.

See Section 11.1 on page 245.

- **Allow Video Calls:** Selecting this option allows the user to perform video calls. Video calls is a user-licensed feature.
- **Allow Telephony Presence:** Telephony presence indicates a user's availability for accepting voice calls. Select this option to allow the user to access telephony presence information about other users.
- **Allow Use of SoftPhone:** Check this box if you want this user to have access to the SoftPhone option in Call Manager.
- **Allow Phone API:** Third-party developers have the ability to create applications that can run on certain ShoreTel IP phone models. (Check with your ShoreTel representative for models that support this behavior.)

Selecting this checkbox enables the IP phone associated with this user to go into PAPI browser mode, thus allowing the phone to run those third-party applications.

- **Allow Mobile Call Manager:** Select this checkbox to enable Mobile Call Manager (MCM) to run on this user's mobile device. The MCM client software consists of a small Java applet that is installed on a mobile device, such as a BlackBerry. The MCM software provides remote/mobile users with an interface similar to ShoreWare PCM.

Clear this checkbox to disable MCM on this user's mobile device.

- **Delayed Ringdown:** Select this checkbox to enable Delayed Ringdown for the extension. If this parameter is not enabled, all other Ringdown parameters are not selectable. Refer to Section 9.10.3.1 on page 174 for information about Ringdown.
 - **Extension:** Select this option to specify a ShoreWare extension as the recipient device, then enter the extension of the recipient device in the corresponding data entry field.
 - **External Number:** Select this option to specify a phone outside of the ShoreWare network as the recipient device, then enter the phone number of the recipient device, including the Trunk Access Code of the ringdown trunk, in the corresponding data entry field.
 - **Ringdown Delay:** Enter the period of time that the phone waits after the user picks up the handset before initiating the ringdown call. Enter 0 in this field to cause the phone to immediately dial the ringdown device whenever the handset is picked u

10.4.4.3 Bottom-Section Parameters

The parameters in the bottom portion of the *Edit User: General* panel, are shown in Figure 10-13.

The screenshot displays the 'Edit User' interface with the 'General' tab selected. The top navigation bar includes buttons for 'New', 'Copy', 'Save', 'Delete', and 'Reset', along with a 'Help' link. Below the navigation bar, there are tabs for 'General', 'Personal Options', 'Distribution Lists', and 'Workgroups', with 'General' being the active tab. A 'Refresh this page' link is also present. The form fields are as follows:

- First Name:
- Last Name:
- Number:
- Mailbox on Server: [Escalation Profiles and Other Mailbox Options](#)
- Accept Broadcast Messages
- Include in System Dial By Name Directory
- Make Number Private
- Fax Support:
- Client Type: Enable Contact Center Integration
- Allow Video Calls:
- Allow Telephony Presence
- Allow Use of Soft Phone
- Allow Phone API
- Allow Mobile Call Manager
- Delayed Ringdown
 - Extension:
 - External Number: (e.g. 9+1 (408) 331-3300)
- Ringdown Delay: sec

At the bottom of the form, there is a copyright notice: © 1998-2008 ShoreTel, Inc. All rights reserved.

Figure 10-13 Edit User: General panel – Bottom section

The following are description of parameters located in the bottom section of the *Edit User: General* panel.

- **Client User ID:** This is the login name that a user uses when logging into the ShoreWare Call Manager or ShoreWare Director. By default, it is the first initial of the user's first name followed by the user's entire last name. This can be changed at your discretion.

This field is automatically filled in when you enter the user's first and last names in the *First Name* and *Last Name* fields.

- **Client Password:** This is the password that a user will use when logging in to the system from the ShoreWare Call Manager or ShoreWare Director. Characters that are entered in this field appear as asterisks.

The default password is changeme. Users are asked to change this password the first time they log in to the system.

It is recommended that you do not change this password because it is used by users who are configuring their Call Manager for the first time.

- **Voice Mail Password:** This is the password that a user uses when logging in to his or her voice mailbox from the telephone. Characters that are entered in this field appear as asterisks.

The default password is 1234. Users are asked to change this password the first time they log into the system.

It is recommended that you do not change this password because it is used by users who are configuring their telephone for the first time.

- **Must Change on Next Login:** This check box is selected by default when a new user is created. This forces the user to enter a new password the first time they log into their mailbox. Once the user has entered a new password, the system clears this checkbox.

If a user has forgotten his or her password, the system administrator can reset this option (i.e. select the checkbox) and enter a generic password, thus allowing the user to re-enter a new password.

- **SIP Password:** Entering a value in the data entry box enables the extension to support SIP. Clearing the SIP Password data fields disables the extension from supporting SIP. Refer to Section 17.2.6.5 on page 341 for more information.
- **Email Address:** This is the user's e-mail address. By default, it is automatically entered when you enter the user's first and last names in the *First Name* and *Last Name* fields. It consists of the first initial of the user's first name followed by the user's entire last name—for instance, sdemont. In addition, the @companyname.com domain is saved in a cookie on your workstation each time you save a user. This information is then presented as a default, which can be changed at your discretion.

Be sure to delete this field if the user does not have or use e-mail.

- **Conference Bridge:** From the drop-down list, choose the conference bridge server to be used for conference calls. The user must also have a class of service that permits conference calls.

- *Server*—Select the conference bridge server to be used by this user.

- *User ID*—Set up a conference call User ID.

- *Password*—Set up a password for this User ID.

- **Instant Message Settings:** Instant Message settings configure the user to use Instant Messaging through Call Manager. Only users with at least Professional Call

Manager can use Instant Messaging. Refer to Section 11.1.3.2 on page 248 for more information.

- Server: This parameter specifies the IM server that services the user's Instant Messaging. The drop-down menu displays the list of available Presence servers.
- User ID: Enter the User ID, as recognized by the specified Presence server, through which the client accesses Instant Messaging functions.
- Password: Enter the password, as recognized by the specified Presence server, that authenticates the User ID to the server.
- **Edit System Directory Record:** This link takes you to the *System Directory Entry* edit page, where you can edit the user's system directory information.

For editing instructions, refer to Section 10.4.11 on page 226.

10.4.5 Personal Options

To configure new and existing users' personal options, click the *Personal Options* tab (shown in Figure 10-14) on the *Edit User* page. After making entries, click the link at the bottom of the page to return to the user's edit page.

Users
Edit User

New Copy Save Delete Reset

► General ▼ **Personal Options** ► Distribution Lists ► Workgroups

User Name: Jane Smith
Extension: 888-55400

Call Control Options:
Current Call Stack Size:
Ring Type:
Automatic Off-Hook Preference: Speakerphone Headset Wireless Headset
 Handsfree Mode
 Call Waiting Tone Enabled
Trunk Group Access Code:
Mailbox for Recorded Calls: Search

Program IP Phone Buttons Copy
Program Personal Call Manager Toolbars
[External Assignment](#)

Call Handling Mode Options:
Current Call Handling Mode: [Delegation](#)
 Outlook Automated Call Handling

Edit Call Handling Modes:
[Standard](#)
[In a Meeting](#)
[Out of Office](#)
[Extended Absence](#)
[Custom](#)

Mailbox Options:
[Find Me](#)
[Escalation Profiles and Other Mailbox Options](#)

Figure 10-14 Edit User Page – Personal Options

10.4.5.1 Parameters

The parameters under the *Personal Options* tab on the *Edit User* page are as follows:

CALL CONTROL OPTIONS

- Current call stack size:** This is the number of calls that can be stacked on a user's extension. When this number is exceeded, calls are either given a busy tone or forwarded, depending on the call handling mode that is currently in effect.

Valid values are 1 through 16 with the limitation that the value specified in the call stack size of the user's CoS becomes the upper limit.
- Ring Type:** Select from the drop-down list of one of the four different ring types. The ring type will apply to ShorePhones. It will not apply to analog telephones.

Note that the ring type is a personal option for the user and not a configuration setting of the phone. When a user moves from phone to phone, their ring type will follow them.

Also, keep in mind that most ShoreTel IP phone models support the ability to load customized ring tones onto each phone, so that each user can have a unique ring tone, if desired. See Section 8.7 on page 120 for details.
- Automatic Off-Hook Preference:** Select the *speakerphone*, *headset*, or *wireless headset* radio button to automatically route incoming or outgoing calls to the desired audio pathway. This option is intended for users who rely primarily on a headset (standard or wireless). This feature eliminates the requirement to manually press the Headset button on the IP phone to activate the headset.

The default is speakerphone.

For configuration instructions, refer to Section 10.4.10 on page 224.
- Handsfree Mode:** This enables the user's handsfree mode. When enabled, dial tone is disabled so that the user can use a headset or speakerphone to answer and make calls from the desktop client.

Handsfree mode is useful for both headset and speakerphone users.

The default is disabled.
- Call Waiting Tone Enabled:** This enables the user's call waiting tone. When enabled, the user hears the tone from the telephone handset when a second call is waiting.

The default is enabled.

Subtle differences can be heard in the call waiting tone one hears for a direct extension as opposed to a monitored BCA extension. This difference has been implemented to help differentiate between the two.
- Trunk Group Access Code:** This lets you assign the default trunk access code for this user. Although this actually lists trunk groups, the only parameter leveraged is the access code. If the user has access to multiple trunk groups with the same access code, these trunks are seized based upon the Network Call Routing feature operation described in the *ShoreTel 8 Planning and Installation Guide*.

This eliminates forcing every user to configure his or her desktop client with the proper access code for dialing.
- Mailbox for Recorded Calls:** Call can be monitored or recorded by people with appropriate permissions. Indicate the mailbox to be used for recordings. The

maximum length of recording is determined by the voice mail class of service for the destination mailbox.

- **Program IP Phone Buttons:** This link takes you to the *Program IP Phone Buttons* page where you can assign functions to the ShorePhone or BB24 programmable buttons.

For configuration instructions, refer to Section 10.4.8 on page 207.

- **Copy:** This button takes you to the *Copy IP Phone Buttons* page. The *Copy* button reduces the tedious work of configuring the programmable buttons on each user's IP phone by allowing a system administrator to copy a programmable button configuration from one user to another.

For configuration instructions, refer to Section 10.4.8.1 on page 209.

- **Program Personal Call Manager Toolbars:** This link takes you to the *Program Personal Call Manager Toolbars* page where you can create new toolbars and assign common functions to the buttons on that toolbar.

The customizable toolbars that extend across the top of the Call Manager user interface. The buttons in these toolbars can be programmed with common operations in a manner similar to the custom buttons on some IP phone models. PCM users can then perform many operations (e.g. "speed dial" or open URL) just by clicking a button on the toolbar.

For configuration instructions, refer to Section 10.4.9 on page 217.

- **External Assignment:** This link takes you to the *Find Me and External Assignment* page where you can set two Find Me numbers as forwarding destinations, or where you can enter an external PSTN phone number for use with the Office Anywhere feature.

For configuration instructions, refer to Section 10.4.7 on page 205 or refer to Appendix A, starting on page 391.

CALL HANDLING MODE OPTIONS

- **Current Call Handling Mode:** This lets you set the current call handling mode for the user. Users can select and edit each call handling mode from their desktop client.

The default is *Standard*.

For more information, see Section 10.9 on page 234.

- **Delegation:** You have the option of delegating the call handling management to one or more other users. This feature is helpful for users who may want their personal assistants to change their call handling mode for them.

For more information, see Section 10.9 on page 234.

- **Outlook Automated Call Handling:** This lets a user's Microsoft Outlook Calendar control his or her call handling mode. Select this check box to enable this feature.

- **Edit Call Handling Modes:** The links under this section bring you to the call handling mode configuration pages. Call handling modes specify how, when, and where calls are forwarded, and whether the user requires message notification when voice mail is received.

For configuration instructions, refer to Section 10.9 on page 234.

MAILBOX OPTIONS

- **Find Me:** This link takes you to the *Find Me and External Assignment* page where you can set two Find Me numbers as forwarding destinations, or where you can enter an external PSTN phone number for use with the Office Anywhere feature.

For configuration instructions, refer to Section 10.4.7 on page 205 or refer to Appendix A, starting on page 391.

- **Escalation Profiles and Other Mailbox Options:** This link takes you to the *Escalation Profiles and Other Mailbox Options* page where you can create *Message Notification Escalation Profiles*, configure *Email Delivery Options* for a user, and set up *Automatic Message Forwarding*.

For configuration instructions, refer to Section 10.10 on page 236.

10.4.5.2 Call Handling Mode Delegation

You can permit call handling changes to be made by other users, such as a personal assistant or receptionist. The users with permission to modify the call handling mode of other users must be using the Operator Call Manager client or the Web Access client.

To delegate call handling:

Step 1 From the *Personal Options* page (see Figure 10-14), click *Delegation*.

The *Call Handling Mode Delegation* page appears, as shown in Figure 10-15.

Step 2 From the left column, select the user(s) to whom call handling will be delegated and click *Add*.

Step 3 To remove a user from call handling delegation, from the right column, select the user you want to remove and click *Remove*.

Figure 10-15 Call Handling Mode Delegation Edit Page

You can filter the users who are displayed on the page. If you have many users in your organization, it may be easier to enter the first letter of the last name, for example, to narrow your search.

10.4.6 Voice Mail Delivery and Notifications

The user's voice mail notification options are edited from the *Voice Mail Delivery and Notifications* page, shown in Figure 10-16. Voice mail notification parameters specify what the user should do if notification is enabled at the time a message arrives. These options include the user's message notification telephone number, pager ID number, and try options.

Voice mail may be auto-forwarded. That is, a mailbox may be configured to send any message it receives to another mailbox. The message sent to the original mailbox can be automatically deleted, as an option. The target mailbox for forwarded messages may be any user, a workgroup, a route point, AMIS address, or a system distribution list (other than a broadcast distribution list). A message is pre-pended to the forwarded message, along with a time-stamp, announcing that the message has been auto-forwarded. As an example, the recipient of an auto-forwarded message might hear, "Auto-forwarded message received at 9:10 AM from Customer Support Mailbox".

An example of use might be the handling of off-hours calls when few support staff are available. Off-hours calls may be routed to a back-up extension. If no one is available to answer the back-up extension, calls may wind up in a voice mailbox that will not be checked for hours. The back-up extension can be set to auto-forward any calls that are received in its mailbox. Calls can be forwarded to a mailbox that is checked on a regular basis.

Auto-forwarding is available between distributed voice mail servers. The message is handled as any other message would be, including any message waiting indicator, any calling notification, return receipt requests, or urgent markings. Auto-forwarded messages can be forwarded and replied to. If the target mailbox is full, the is left in the sending mailbox. If a message is being auto-forwarded to a list of mailboxes and one is full, that target is skipped.

You can get to this edit page by clicking the *Voice Mail Delivery and Notifications* link under the *Personal Options* tab on the *Edit User* page.

- **Parameters:** The parameters on the *Voice Mail Delivery and Notifications* page are as follows:
- **Email Delivery:** Select one of three e-mail delivery options: *Disabled*, *Email Notification*, or *Email Copy of VM*. The Email Notification option notifies the user of the time, duration, and Caller ID of the message that was recorded. The Email Copy of VM options attaches the voice message to the notification email as a WAV file.
- **Email Address:** The email notification can be configured to be delivered to any SMTP address.
- **Marked Delivered Voice Mail as Heard:** The notification can be configured to mark the original message as heard to prevent other forms of notification taking place and turns off the message waiting indicator (MWI).
- **Automatic Message Forwarding Destination:** Messages may be auto-forwarded to a user, a workgroup, a route point, or a system distribution list. If the system distribution list includes AMIS destinations, they also receive the auto-forwarded message.

Voice Mail Delivery and Notifications

Adam Ant

[Help](#)

Edit this record
[Refresh this page](#)

Email Delivery:

Email Address:

Mark Delivered Voice Mail as Heard

Automatic Message Forwarding:

Destination: None Extension: AMIS:

Delete Message after Forwarding

Notification by Phone:

Voice Mail Notification Condition: All Urgent Only None

Voice Mail Notification Method: Pager Phone

Notification Number: Extension: External: (e.g. +1 (408) 331-3300)

Pager ID:

Pager Data:

Try Count:

Try Interval: minutes

[Return to this user](#)

Figure 10-16 User's Voice Mail Delivery and Notifications Page

The default is None, meaning, no forwarding.

The destination may not be a broadcast distribution list.

- **Delete Message After Forwarding:** Check the checkbox to cause the automatic deletion of the message after forwarding. The default is not to delete.
- **Notification By Phone – Voice Mail Notification Condition:** These options let you select the types of voice mail messages that trigger a notification. Voice mail notification is set in Section 10.2.4 on page 183.
 - *All* notifies the user the first time a message is received.
 - *Urgent Only* notifies the user the first time an urgent message is received.
 - *None* does not notify the user.
- **Voice Mail Notification Method:** These options let you specify how voice mail messages are received, either by pager or by phone.
- **Notification Number:** This is the number where message notifications are directed. It must contain any applicable and long-distance dialing sequence, but must not include an access code.
- **Pager ID:** If the notification method is *Pager*, this is the pager identification number (PIN) that the paging service provided.
- **Pager Data:** This is the number that the user wants displayed by the pager to indicate that a message is waiting.

It is recommended that you test pagers with your pager service to ensure proper functionality.

- **Try Count:** This is the number of times that the user wants the system to try message notification. The ShoreTel 6.1 system continues to call the user for the number of times indicated in this parameter, with the number of minutes between each try as configured in the *Try Interval* parameter.

The user can stop the system from trying simply by logging in to the voice mail system.

- **Try Interval:** This is the number of minutes that the voice mail system waits between attempts to notify the user.

10.4.7 Find Me and External Assignment

When you click the *Find Me and External Assignment* link on the *Personal Options* page, the *Find Me and External Assignment* page appears, as shown in Figure 10-17.

Find Me and External Assignment
Jane Smith

[Save](#) [Reset](#)

[Edit this record](#) [Refresh this page](#)

Find Me:

Primary Destination:

None

Extension: [Search](#)

External: (e.g. 9 +1 (408) 331-3300)

Number of Rings:

Backup Destination:

None

Extension: [Search](#)

External: (e.g. 9 +1 (408) 331-3300)

Number of Rings:

Send Incoming Caller ID

Enable Auto Find Me

External Assignment:

Use External Assignment

Phone Number: (e.g. 9 +1 (408) 331-3300)

Activation:

Figure 10-17 Find Me and External Assignment Page

Find Me allow users to configure two phone numbers as forwarding destinations, so that if they miss an incoming call, the call will be sent to voice mail, and from there, it will be redirected to one of the pre-configured forwarding destinations. **External Assignment** is also known as “Office Anywhere.” This feature offers a way to forward calls to an off-system (PSTN) extension (see Section A.3 on page 393 for information on configuring External Assignment).

10.4.7.1 Find Me Destinations

When a caller dials a user on the ShoreTel 6.1 system, if the call is not answered, it is sent to voice mail. At this point, the caller is offered the option to “press 1 to activate Find Me call handling,” and the system subsequently attempts to route their call to one of the Find Me destinations that the user configured.

There is no system prompt informing the caller that they need to press 1. The ShoreTel user must inform callers of this “press 1 to activate Find Me” option in the outgoing voice mail greeting.

When callers presses “1”, they hear a prompt telling them that the Find Me destinations are being called. If the call is not accepted at either of the Find Me destinations, the call is sent to voice mail.

Alternatively, users also have the option to automate the Find Me behavior, thus bypassing the requirement for callers to “press 1 to activate Find Me.” When **Auto Find Me** is enabled, calls will be immediately sent to the Find Me destination number(s) without any action on the part of the caller.

Details:

- Find Me destinations can be extensions or external numbers.
- Find Me call handling can be enabled/disabled for each of the five call handling modes. The same Find Me destinations apply to all call handling modes with Find Me enabled.

To configure Find Me destinations:

Step 1 For the primary Find Me destination, select *Extension* or *External*. You can use the *Search* button to find system extensions. Enter external numbers in the *External* text box.

Step 2 Enter a value in the *Number of Rings* field. This is the number of times the phone must ring without being answered before the call is forwarded to the primary Find Me destination.

Step 3 For the *backup* Find Me destination, select *Extension* or *External*. You can use the *Search* button to find system extensions. Enter external numbers in the *External* text box.

Step 4 Once again, enter a value in the *Number of Rings* field. This is the number of times the phone must ring without being answered at the primary Find Me destination before being forwarded to the *backup* Find Me destination.

Step 5 Select the *Send Incoming Caller ID* check box if you want the caller ID forwarded to the Find Me destination.

Step 6 Select the *Enable Auto Find Me* check box to automate the process of routing calls to the Find Me destinations. With this enabled, users will no longer have to press 1 to activate Find Me. You can deselect the check box if you wish to continue to require callers to “press 1 to activate the Find Me feature.”

If you enable *Auto Find Me*, make sure that your outgoing voice mail message is no longer telling callers to “press 1 to activate the Find Me feature.” The automation aspect of this enhancement means that callers will not have to do anything and calls will be forwarded automatically.

Step 7 Click *Save* to store your changes.

10.4.8 Programmable IP Phone Buttons

An administrator or user can change the functions associated with the ShorePhone or BB24 custom buttons. Users can create shortcuts for operations that would normally require pressing two or three buttons to accomplish the same task.

The action associated with the bottom button on an IP 530/560 could be configured to speed dial a particular extension or external number. The button above that could be set to perform overhead paging, and so on. All of the custom buttons are configurable except for the top-most button, which is permanently set to provide call appearance information (i.e. ringing indicator and call timer information).

Users can keep track of which actions they perform on a regular basis and then associate those actions with the custom buttons. Instead of having to dial a star code (such as *14 for picking up the Night Bell) they can just press one button. See Table 10-1 for supported functions.

Details:

- After a function is assigned to a button, users can enter a label (up to five characters on multiline ShorePhones and up to six characters on the IP 100 and BB24). The label appears on the LED display next to the custom button, except on IP100 phones, where the label appears on the soft key.
- The system administrator can configure the custom buttons via ShoreWare Director on behalf of a user, or he can enable permissions for an individual user so that the user can modify the custom buttons on his own IP phone via the telephone interface.
- The programmable button feature is supported on the IP100 and multiline models but is not supported on the analog models or on the IP 210.
- The IP100 does not have custom buttons, but the soft keys can be configured (via ShoreWare Director) in the same way that custom buttons are configured on the other phones. IP 100 phones cannot be configured by the user. The buttons on these phones must be configured by the system administrator.

Function	Parameter	Comments
Barge In	Extension or none	
Bridged Call Appearance	Extension or none	
Call Appearance	None	Not supported on Button Box
Centrex Flash	None	Refer to Section 7.2.10 on page 107.
Conference Blind	Extension or external number	
Conference Consultative	Extension or external number	
Conference Intercom	Extension or none	
Dial Mailbox	Extension or none	
Dial Number (Speed Dial)	Extension or external number	
Intercom	Extension or none	
Monitor Extension	Extension or none	
Page	None	
Park	Extension or none	
Park and Page	Extension or none	

Table 10-1 Supported Programmable Button Functions:

Function	Parameter	Comments
Pickup	Extension or none	
Pickup Night Bell	None	
Pickup/Unpark	Extension or none	Uses internal presence to determine which operation to perform
Record Call	Mailbox	Operates on selected call in connected; Pressing a second time stops the recording. Call recordings can be saved in the mailbox of the initiating client (by leaving the mailbox field blank), or can be routed to an alternate mailbox by entering a mailbox number in the field.
Record Extension	Extension or mailbox	Call recordings can be saved in the mailbox of the initiating client (by leaving the mailbox field blank), or can be routed to an alternate mailbox by entering a mailbox number in the field.
Send Digits Over Call	Extension	
Silent Monitor	Extension or none	
Transfer Blind	Extension or external number	
Transfer Consultative	Extension or external number	
Transfer Intercom	Extension or none	
Transfer to Mailbox	Extension or none	
Transfer Whisper	Extension or none	
Unpark	Extension or none	
Whisper Page	Extension or none	
Whisper Page Mute	None	

Table 10-1 Supported Programmable Button Functions:

Configuring Programmable Buttons via Director

To configure programmable buttons via ShoreWare Director, follow the procedure below:

- Step 1** Launch **ShoreWare Director** and enter the user ID and password. Then click the *Login* button.
- Step 2** Click on the *Administration* link to expand the list (if it has not already been expanded).
- Step 3** Click on the *Users* link and then the *Individual Users* link to display the list of users on this system.
- Step 4** Click on the name of the user whose phone you want to modify and then click on the *Personal Options* tab.
- Step 5** Scroll to the bottom of the window and click on the *Program Buttons* link to display a window similar to the one shown below:
- Step 6** Click on the *Function* drop-down menu and select the action you would like to associate with a particular button.
- Step 7** In the *Label* field to the right of the drop-down menu, enter a descriptive word to remind the user which function is associated with that button. This label will

Customize IP Phone Buttons [Save](#) [Reset](#) [Help](#)
 Adam Ant * modified

Edit Custom Keys [Refresh this page](#)

IP Phone Type:

Button	Function	Label	Target
1	Call Appearance		
2	Transfer Blind	Blind	<input type="radio"/> None <input checked="" type="radio"/> Extension: <input type="text" value="818-5302 : HQ IP2"/> Search <input type="radio"/> External: <input type="text"/> (e.g. 9+1 (408) 331-3300)
3	Intercom	Intcm	Extension: <input type="text"/> Search
4	Page		
5	Unpark		Extension: <input type="text"/> Search
6	Page		

[Return to](#)

- Page
- Park
- Park and Page
- Pickup
- Pickup Night Bell
- Pickup/Unpark
- Record Call
- Record Extension
- Send Digits Over Call
- Silent Monitor
- Transfer Blind

Figure 10-18 Programming Custom Keys on IP 530 Phone

appear on the LED display adjacent to the button. Note that this label can be up to five characters long for the multiline phones and up to six characters long for the IP 100 and BB24.

Step 8 After you have selected a function and entered a label for a particular button, you may optionally enter destination information in the *Target* field. Certain functions do not require entering a destination, but other functions such as speed-dial or blind transfer can optionally take a destination. Some functions take only extensions and some take any type of phone number.

Step 9 When you have finished entering functions and labels for all of the buttons, click the *Save* button at the top of the window to store your changes.

10.4.8.1 Copying Programmable Buttons Configurations

The *Copy IP Phone Buttons* page allows a system administrator to copy the programmable button configuration from one use to another, thus reducing the tedious work of configuring IP phone buttons.

The following IP phone models support this functionality: IP100, BB24 and multiline devices.

To copy the programmable button configuration from one user's IP phone to another, 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 *Users* link and then the *Individual Users* link to display the list of users on this system.
- Step 4** Click on the name of the user whose IP phone you would like to modify, and then click the *Personal Options* tab to display a window similar to the one shown below:

The screenshot shows the 'Users' administration page for 'Jane Smith'. At the top, there are buttons for 'New', 'Copy', 'Save', 'Delete', and 'Reset'. Below these are four tabs: 'General', 'Personal Options' (which is selected and highlighted in orange), 'Distribution Lists', and 'Workgroups'. Under the 'Personal Options' tab, the following fields are visible:

- User Name: Jane Smith
- Extension: 888-55400
- Call Control Options:**
 - Current Call Stack Size: 4
 - Ring Type: Standard
 - Automatic Off-Hook Preference: Speakerphone Headset Headset with Electronic Lifter
 - Handsfree Mode
 - Call Waiting Tone Enabled
 - Trunk Group Access Code: <None>
 - Mailbox for Recorded Calls: [text field] Search
- Program IP Phone Buttons: Copy
- Program Personal Call Manager Toolbars
- External Assignment

Figure 10-19 Accessing the *Copy* button via the *Personal Options* tab

- Step 5** Click on the *Copy* button to display a window similar to the one shown below:

The screenshot shows a pop-up dialog window titled 'Copy IP Phone Buttons -- Web Page Dialog'. The window is for user 'Jane Smith'. It contains the following elements:

- From User:** 3333-56789 : Kim Waya Search
- Phone Type:**
 - IP100
 - IP212k/230/530/560/560g
 - BB24 Button Box 1
 - BB24 Button Box 2
 - BB24 Button Box 3
 - BB24 Button Box 4
- Buttons: Copy, Cancel
- Footer: © 1998-2006 ShoreTel, Inc. All rights reserved.
- Address bar: http://10.23.26.100/shorewaredire Internet

Figure 10-20 Copy IP phone buttons pop-up window

- Step 6** Click the *Search* button next to the “From User” field to locate the person whose programmable button setup you wish to copy.

Step 7 Select the appropriate *Phone Type* check box. The phone types must match, meaning that you cannot copy an IP100 button configuration to an IP230.

If both users have multiple phones on their desk (e.g. an IP560 phone with a BB24 button box), you can copy the button configuration for several devices simultaneously, assuming the other user also has an IP560 phone and a BB24 button box). To do this, you would just select the appropriate Phone Type check boxes. In this manner, you could copy up to four button box configurations at once.

Step 8 Click the *Copy* button to duplicate the programmable button configuration from one user to another.

To enable a user (as opposed to a system administrator) to configure the programmable buttons on his or her ShorePhone, 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 **Users** link and then the **Individual Users** link to display the list of users on this system.
- Step 4** Click on the name of the user whose profile you would like to modify (enabling him to customize the buttons on his IP phone). A window similar to the one shown below appears.

Users
Edit User

[New](#) [Copy](#) [Save](#) [Delete](#) [Reset](#)

▼ **General** ▶ Personal Options ▶ Distribution Lists ▶ Workgroups

First Name:

Last Name:

Number:

License Type:

Caller ID: (e.g. +1 (408) 331-3300)

DID: (DID Range: +14083313921 - 3940)

PSTN Failover:

User Group: [Go to this User Group](#)

Figure 10-21 Accessing the User Group via the Edit User window

- Step 5** Click on the **Go to this User Group** link and on the next window that appears, click on the **Go to this Class of Service** link located to the right of the COS - Telephony field. The Class of Service window appears, similar to the one shown below.
- Step 6** Enter a name for the Class of Service profile in the **Name** field (unless you are modifying an existing Class of Service profile for which you would like to preserve the existing name).
- Step 7** Select the **Allow Customization of IP Phone Buttons** check box.
- Step 8** Click the **Save** button to store your changes.

Details:

- The default for new Class of Service (Telephony) profiles is to have this feature disabled, thus preventing users from modifying their own custom buttons.
- Once this Class of Service (Telephony) profile has been created, you can apply it to every person in the system that you would like to allow to change his or her custom buttons.

To change the custom buttons on your ShoreP\\ phone or BB24 via the telephone interface, follow the procedure below:

Class of Service
Edit Telephony Features Permissions

[New](#) [Copy](#) [Save](#) [Delete](#) [Reset](#) [Help](#)

[Edit this record](#) Refresh this page * modified

Name:

Max. Call Stack Depth:

Max. Parties in Make Me Conference:

Allow Call Pickup

Allow Trunk-to-Trunk Transfer

Allow Overhead and Group Paging

Allow Make Hunt Group Busy

Allow Extension Reassignment

Allow PSTN Failover

Show Caller ID Name and Number on Monitored Extensions

Allow Customization of IP Phone Buttons

Show Extensions with Different Prefixes in Directory

Allow Collaboration Features

Allow Recording of Own Calls

Directed Intercom / Paging: Allow Initiation

Figure 10-22 Enabling customization of IP phone buttons

- Step 1** Press the *Options* button on your IP phone and enter your password, followed by the # key.
- Step 2** Scroll through the list to *option 4. Program Buttons*.
- Step 3** Press the *Edit* soft key.
- Step 4** Press the custom button that you would like to modify. (If you are using the phone interface to modify the buttons on a BB24 device, press the button on the BB24 that you would like to configure.)
- Step 5** Scroll through the list of functions until you find the function that you would like to apply to this button.
- Step 6** When you have highlighted the appropriate function, press the *Next* soft key.
- Step 7** Enter an extension, external number, or leave it blank. Then, press the *Next* soft key.
- Step 8** Press the *1 -> Aa* soft key to shift the key pad to alphabet mode.
- Step 9** Use the key pad to enter a short descriptive word that will remind you of the new function of the custom button.
- Step 10** Press the *Done* soft key.
- Step 11** Press the *Done* soft key again to store your changes.

10.4.8.2 Monitor Extensions from IP Phone

Extension monitoring is one of the available functions you can assign to the custom buttons on ShorePhone. However, due to its complexity the feature will be discussed separately from the other functions.

The extension monitoring function allows a user to monitor the extension of another user and to answer that person's calls if necessary. For example, consider a hypothetical scenario in which there are two secretaries working on different floors of the same building. They are both responsible for answering calls from the main phone line but they cannot see one another. If one of them is already on a call and another call arrives on the main line, with extension monitoring enabled, the second assistant will see that the first assistant is busy and will know to pick up the incoming call.

To configure extension monitoring, follow the directions below:

- Step 1** Launch **ShoreWare Director** and enter the user ID and password. Then click the **Login** button.
- Step 2** Click on the **Administration** link to expand the list (if it has not already been expanded).
- Step 3** Click on the **Users** link and then the **Individual Users** link to display the list of users on this system.
- Step 4** Click on the name of the user whose phone you would like to modify and then click on the **Personal Options** tab.
- Step 5** Scroll to the bottom of the window and click on the **Customize IP Phone Buttons** link.
- Step 6** Click on the **Function** drop-down menu for one of the programmable buttons and select **Monitor Extension** to display a window similar to the one in Figure 10-23.

Customize IP Phone Buttons Save Reset
Adam Ant

Edit Custom Keys Refresh this page

IP Phone Type: IP530/560

Button	Function	Label	Target
1	Call Appearance		
2	Monitor Extension		Extension: 82-110 : 110 Search Ring Delay Before Alert: 4 Caller ID on Monitored Extensions: <input type="radio"/> Do Not Show <input checked="" type="radio"/> Show Only When Ringing <input type="radio"/> Show Always No Connected Call Action: Dial Number (Speed Dial) With Connected Call Action: Transfer Blind
3	Call Appearance		
4	Call Appearance		
5	Call Appearance		
6	Call Appearance		

[Return to this user's personal options](#)

Figure 10-23 Monitor Extension from IP Phone

- Step 7** Enter a label for the button in the *Label* field.
- Step 8** Enter the extension of the monitored party in the *Extension* field, or click the *Search* button to locate the desired extension.
- Step 9** Click on the *Ring Delay Before Alert* drop-down menu and select the number of rings that should elapse on the monitored party's phone before the monitoring party will receive an alert. (This delay gives the monitored party a chance to answer and prevents the monitoring party's phone from ringing incessantly.)
- Step 10** Select the appropriate radio button in the *Caller ID on Monitored Extensions*. For an multiline phone, choices are:
- **Do Not Show** – The Caller ID information does not appear, but an indicator will show that the monitored phone is busy. This option offers the monitored party the most privacy and should be selected if you do not want the monitoring party to know who the monitored party is talking to.
 - **Show Only When Ringing** – The Caller ID information appears while the phone is ringing, but disappears once the call has been answered.
 - **Show Always** – The Caller ID information appears while the monitored phone is ringing, and continues to appear even after the call has been answered.
- Step 11** The custom buttons can be configured to perform different actions based on whether or not the person being monitored is on a call. To associate a secondary function with the custom button that will apply when the phone is inactive, click the drop-down menu and select another action from the *No Connected Call Action* menu. The action you select here will apply when the custom button is pressed AND while the monitoring party's phone is inactive.
- Step 12** To associate a third function with the custom button, click the drop-down menu and select another action from the *With Connected Call Action* menu. This action will apply when the custom button is pressed while the monitoring party does not have a call that can be picked up or unparked, and the user's own extension has a connected call.
- The idea behind these last two steps is that the custom button can be configured to do whatever makes sense given the situation. The button can be configured to pick up incoming calls, or park/unpark calls when the person being monitored is on a call. And if the person being monitored is not on a call, the custom button then becomes a one-touch button allowing the operator to transfer a call to the monitored extension. And if the person being monitored is not on a call and if there are no calls to transfer, then the custom button becomes a speed-dial button, allowing the operator to dial that person's extension at the touch of a button.
- Step 13** Click *Save* to store your changes.

Details:

- The custom button will illuminate red (on the monitoring-person's phone) when the person being monitored is on a call. If that call is put on hold and a second call is accepted on the monitored extension, the LED will turn green and will flash twice. Similarly, the LED will flash three times if a third call is accepted. For more information about LED flash patterns, see Table 10-1.

- The custom button (to which extension monitoring has been assigned) can serve dual purposes based on whether the monitoring party is in a call or not. The button can be set to speed dial, intercom, or transfer calls to the monitored extension.
- When the "Show Caller ID Name and Number on Monitored Extensions" Class of Service (Telephony) setting is not enabled, PCM Extension Monitor (and Agent Monitor) show the number of calls on a user's stack but do not show who the user is talking to. "Properties" is also disabled.

State	Pattern
CALL APPEARANCE STATES	
Idle	Off
Idle and DND	Orange, Steady On
Idle and Message Waiting	Off
Idle, Message Waiting and DND	Orange, Steady On
Off Hook	Green, Steady On
Active Call	Green, Steady On
Active Conference Call	Green, Steady On
Remote Hold	Green, Steady On
Offering Call	Green, 1000/1000 ms
Held or Parked Call (3)	Orange, 250/250 ms
Whisper Page Call	Red, Steady On
Active Call Whisper Muted	Red, Steady On
EXTENSION MONITOR STATES	
Idle	Off
Idle and DND	Orange, Steady On
Idle and Message Waiting	Off
Idle, Message Waiting and DND	Orange, Steady On
Offering Call	Green, 1000/1000 ms
Active Call Picked Up	Green, Steady On
Held or Parked Call [3]	Orange, 250/250 ms
Monitored Ext. on Active Call	Red, Steady On
Monitored Ext. on Conference Call	Red, Steady On
Monitored Ext on Active Call + Offering Call	Green, 200/100/700/1000 ms
Picked up Monitored Ext. Call + Monitor Ext on Active Call	Green, 800/Orange 200 ms
Picked up Monitored Ext. Call and Held + Monitor Ext on Active Call	Orange, 200/100/200/500 ms
Picked up Monitored Ext. Call + Monitor Ext held Active Call	Orange, 200 ms Green, 800 ms Orange, 200 ms Green, 100 ms
BRIDGED CALL APPEARANCE STATES	
Idle	Off
Offering Call	Green, 1000/1000 ms
Active Call Picked Up	Green, Steady On

Table 10-2 Programmable Buttons LED Flash Patterns

State	Pattern
Line In-Use	Red, Steady On
Held or Parked Call [3]	Orange, 250/250 ms
FEATURE KEY WITH EXTENSION TARGET STATES	
Idle or Offering Call	Off
Connected or Held Call	Red, Steady On
DND	Orange, Steady On
(Dial/Transfer Mailbox Only) MWI	Red, Steady On
(Pickup, Pick/Unpark, Pickup NightBell Only) Offering	Green, 1000/1000 ms
(Unpark, Pick/Unpark Only) Held/Parked	Orange, 250/250 ms
TOGGLE FUNCTIONS (RECORD, WHISPER MUTE)	
Function Off	Off
Function Available	Orange, Steady On
Record Active	Orange, 500/500 ms
Whisper Mute Active	Orange, 500/500 ms

Table 10-2 Programmable Buttons LED Flash Patterns

10.4.9 Programming Personal Call Manager Toolbar

Programmable Toolbars for PCM are similar to regular toolbars in that they extend across the top of the Call Manager user interface. However, the buttons in these new toolbars can be programmed with common operations in a manner similar to the programmable buttons on some IP phone models.

Once a user's toolbar buttons have been configured, the user can perform many basic telephony operations just by clicking a button in PCM. For example, a button could be configured to "speed dial" another extension or open up the user's default browser to a programmed URL when clicked.

In addition, Programmable Toolbars also provide the foundation for integrating ShoreTel PCM with Contact Center's Agent Toolbar. End users can assign Contact Center functions to the buttons on the PCM Toolbar and control their contact center state while accessing call control functions from a single, unified interface.

Details:

- Toolbars must be configured by the system administrator via ShoreWare Director.
- Toolbars can be configured on a per-user basis (see Table 10.4.9.1 on page 219) or on a global basis (see Section 10.4.9.2 on page 222.)
- Up to 6 toolbars can be defined per user, with each toolbar supporting up to 24 programmable buttons.
- Each user can additionally inherit up to 3 global toolbars through their user group.
- Each toolbar can exist on a separate row in the UI or toolbars can share a row.
- Rows can be docked or moved around the top portion of the window as a group.
- Individual toolbars can be shown or hidden from the View menu in the Call Manager user interface (see Displaying Toolbars on page 72).

- Buttons may combine an operation with a parameter (such as a user extension), allowing one-click access to commonly performed operations, such as blind-transferring to a particular user.
- Buttons assigned with a particular operation will be disabled when their corresponding menu items are disabled. For example, some of the Contact Center function buttons are not available as menu items, thus ensuring that the system administrator can easily block access to those functions on a per-user basis.
- Buttons that are associated with an extension (such as Blind Transfer or Extension Monitor) will show the PCM-user when the monitored party has call activity. Additional presence information appears if the user hovers the cursor over the associated button.
- The user can have any and all toolbars active at once.
- Supported operations are listed on the following page.

Supported operations:

- Add/Modify Contact
- Agent Login
- Agent Logout
- Agent Wrap-Up
- Answer
- Answer Call Center Call
- Assign to Last External Number (Office Anywhere / Extension)
- Barge In
- Blind Transfer Agent
- Bridged Call Appearance
- Change CHM
- Change Default Audio Path
- Conference
- Conference Blind
- Conference Consultative
- Conference Intercom
- Consult Transfer Agent
- Dial Mailbox
- Dial Number (Speed Dial)
- Edit Call Note
- End Wrap-Up
- Execute DDE command
- Extend Wrap-Up
- Go Home
- Group Pickup
- Hangup
- Help
- Hold
- Intercom
- Invoke Command line
- Invoke URL
- Login Group
- Login Primary Groups
- Logout Group
- Logout Primary Groups

- Monitor Extension
- Open Agent Monitor
- Open Conference Mgr
- Open Control Panel
- Open Directory
- Open Extension Monitor
- Open External Assignment
- Open History Viewer
- Open Queue Monitor
- Open Soft Phone
- Open Voice Mail
- Page
- Park
- Park and Page
- Pickup
- Pickup Night Bell
- Record Call
- Record Extension
- Reinsert Busy Call
- Reinsert Terminated Call
- Reinsert Unanswered Call
- Release with Code
- Resume/Release
- Run Contact Center App
- Send Digits Over Call
- Set Agent ID
- Silent Monitor
- Supervisor Help
- To AA
- To VM
- Toggle Handsfree
- Transfer
- Transfer Blind
- Transfer Consultative
- Transfer Intercom
- Transfer to Mailbox
- Transfer Whisper
- Unpark
- Whisper Page
- Whisper Page Mute
- Wrap Up Code

10.4.9.1 Creating a Personal Programmable Toolbar

To configure the buttons of a user's Call Manager Programmable Toolbar, follow the instructions below:

- Step 1** Launch ShoreWare Director and enter the user ID and password. Then click the *Login* button.
- Step 2** Click on the *Administration* link to expand the list (if it has not already been expanded).

Step 3 Click on the *Users* link and then click on the *Individual Users* link.

Step 4 Click on the name of the user whose toolbar you would like to configure. A window similar to the one shown below appears:

Users
Edit User

[New](#) [Copy](#) [Save](#) [Delete](#) [Reset](#)

► **General** ▼ **Personal Options** ► Distribution Lists ► Workgroups

User Name: Jane Smith
Extension: 888-55400

Call Control Options:

Current Call Stack Size:

Ring Type:

Automatic Off-Hook Preference: Speakerphone Headset Headset with Electronic Lifter

Handsfree Mode

Call Waiting Tone Enabled

Trunk Group Access Code:

Mailbox for Recorded Calls: [Search](#)

Program IP Phone Buttons [Copy](#)

[Program Personal Call Manager Toolbars](#)

[External Assignment](#)

Figure 10-24 Edit Personal Options window

Step 5 Click the *Program Personal Call Manager Toolbars* link to display a window similar to the one shown below:

Program Personal Call Manager Toolbars [Help](#)

888-55400

User Toolbar List 0 records checked.

[Delete](#) [New...](#)

Name

[Return to this user's personal options](#)

Figure 10-25 Program Personal Call Manager Toolbars window

Step 6 Click the *New* button to display a window similar to the one shown below:

Step 7 Enter a name for this toolbar in the *Name* field and then click on the first *Function* drop-down menu (which should say "All") and select the category. Options are:

- **All** - This lists all operations
- **Contact Center** - (E.g. Login Group, Logout Group, Answer Call Center Call, etc.)
- **Config** - (E.g. toggle Handsfree mode, toggle audio path, Agent Wrap-Up, etc.)
- **Other** - (E.g. Unused)
- **Telephony** - (E.g. Conference, Intercom, Group Pickup, etc.)

Program Personal Call Manager Toolbar Buttons New Copy Save Delete Reset

Edit Toolbar Buttons [Refresh this page](#)

Name:

#	Function	Label	Target
1	Config <input type="text" value="Change CHM"/>	<input type="text" value="mtg"/>	Call Handling Mode: <input type="text" value="In a Meeting"/>
2	Telephony <input type="text" value="Barge In"/>	<input type="text"/>	Extension: <input type="text" value="888-55678 : Gabrielle Reece Constan"/> <input type="button" value="Search"/>
3	Windowing <input type="text" value="Open Control Panel"/>	<input type="text"/>	
4	All <input type="text" value="Unused"/>	<input type="text"/>	
5	All <input type="text" value="Unused"/>	<input type="text"/>	
6	Contact Center <input type="text" value="Unused"/>	<input type="text"/>	
7	Other <input type="text" value="Unused"/>	<input type="text"/>	
	Telephony <input type="text" value="Unused"/>	<input type="text"/>	
	Windowing <input type="text" value="Unused"/>	<input type="text"/>	
	All <input type="text" value="Unused"/>	<input type="text"/>	

Figure 10-26 Configuring a new toolbar

- **Windowing** - (E.g. Open History Viewer, Open Agent Monitor, Open Control Panel, etc.)

If you are configuring the integrated PCM/Contact Center, you must select *Contact Center* from the left-most *Function* drop-down menu, as shown in Figure 10-26.

Step 8 Click on the second *Function* drop-down menu (which should say "Unused") and select the specific operation this button will perform.

Step 9 Enter a label in the *Label* field. The maximum limit for the label is 12 characters.

Step 10 In the *Target* field, enter the appropriate information as required for the type of operation the button will be performing. (For example, if the desired operation is changing the CHM, then a drop-down menu would appear in the Target area, allowing you to select from the various modes.)

Parameters are mandatory for some functions (e.g. Change CHM) and are optional for others (e.g. Blind Transfer). For optional parameters, the user is prompted to provide information for the optional parameter the first time the function executed via the programmable toolbar.

Step 11 Continue programming the toolbar buttons as desired.

If a button is left unused or blank and this unused button is between other used buttons, PCM will interpret the sequence of blank buttons as a divider that will be visible on the toolbar.

Step 12 Once you have finished configuring the toolbar programmable buttons, click *Save* to store your changes.

When finished programming the toolbar for a user, the user's Call Manager window should now have a toolbar similar to the one shown below:

A toolbar programmed with Contact Center functions will appear as shown below:

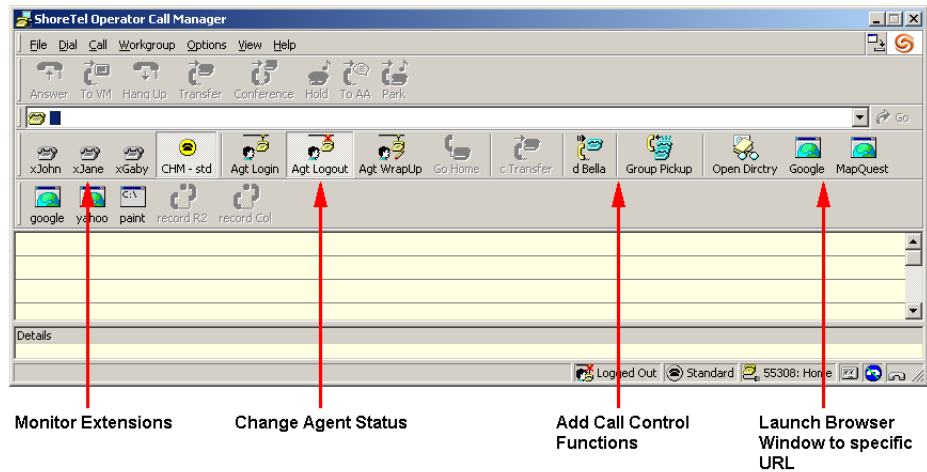


Figure 10-27 Call Manager window with programmable toolbar+

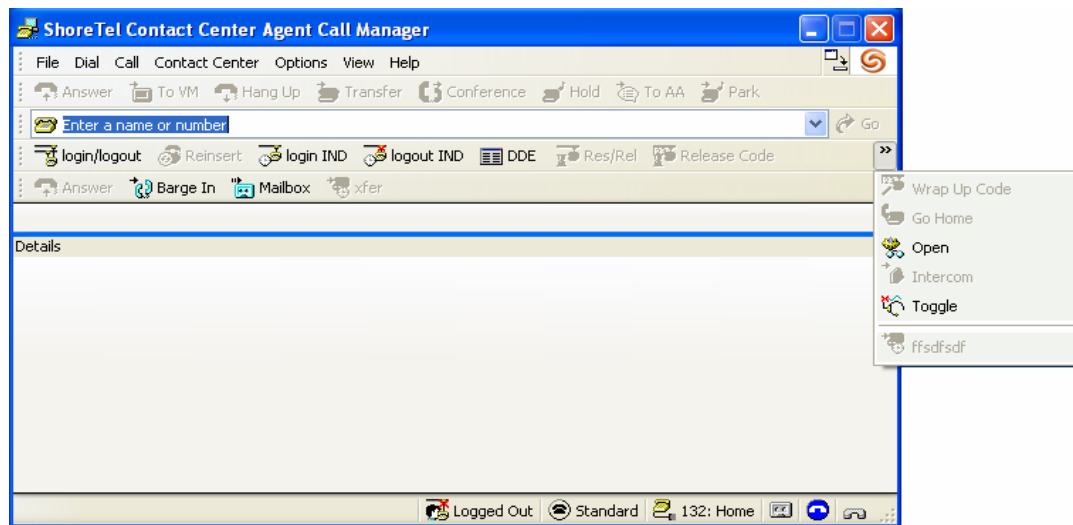


Figure 10-28 Integrated PCM with Contact Center functions

10.4.9.2 Creating a Global Programmable Toolbar

Alternatively, instead of configuring the Programmable Toolbar for an individual user, you can configure a global Programmable Toolbar which can then be applied to many users at once, thus simplifying the configuration process.

To create a global Programmable Toolbar, follow the instructions below:

- Step 1** Launch ShoreWare Director and enter the user ID and password. Then click the *Login* button.
- Step 2** Click on the *Administration* link to expand the list (if it has not already been expanded).
- Step 3** Click on the *Personal Call Manager* link and then click on the *Global Toolbars* link to display a window similar to the one shown below:



Figure 10-29 Creating a new global programmable toolbar

Step 4 Click the **New** button to display a window similar to the one shown below:

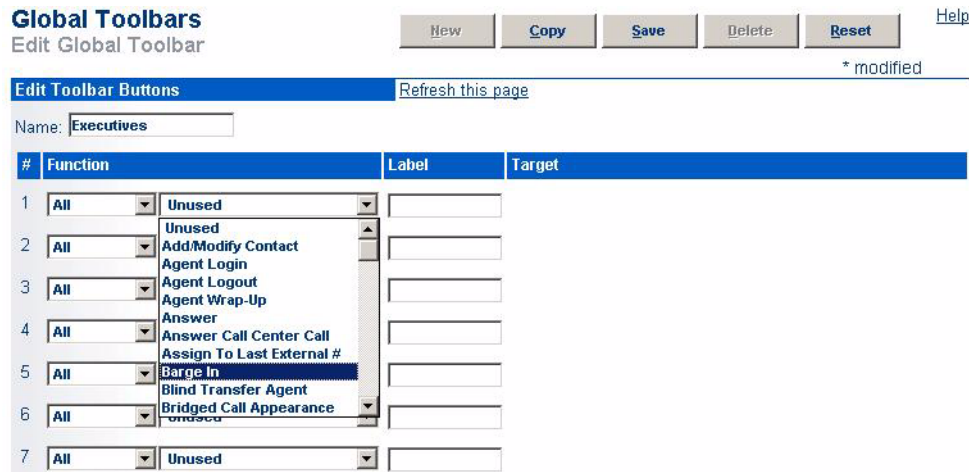


Figure 10-30 Editing global toolbar

Step 5 Enter a name for this toolbar in the *Name* field.

Step 6 Click on the first *Function* drop-down menu (which should say "All") and select the category. Options are:

- **All** - This lists all operations
- **3rd Party** - (E.g. Login Group, Logout Group, Answer Call Center Call, etc.)
- **Config** - (E.g. toggle Handsfree mode, toggle audio path, Agent Wrap-Up, etc.)
- **Other** - (E.g. Unused)
- **Telephony** - (E.g. Conference, Intercom, Group Pickup, etc.)
- **Windowing** - (E.g. Open History Viewer, Open Agent Monitor, Open Control Panel, etc.)

Step 7 Click on the second *Function* drop-down menu (which should say "Unused") and select the specific operation this button will perform.

Step 8 Enter a label in the *Label* field. The maximum limit for the label is 12 characters.

Step 9 In the *Target* field, enter the appropriate information as required for the type of operation the button will be performing. (For example, if the desired operation is changing the Call Handling Mode, then a Call Handling Mode drop-down menu would appear in the Target area, allowing you to select from the various CHM's.)

Parameters are mandatory for some functions (e.g. Change CHM), while they are optional for others (e.g. Blind Transfer). For optional parameters, the user is prompted to provide information for the optional parameter the first time the function executed via the programmable toolbar.

Step 10 Continue programming the toolbar buttons as desired.

If a button is left unused or blank and this unused button is between other used buttons, PCM will interpret the sequence of blank buttons as a divider that will be visible on the toolbar.

Step 11 Once you have finished configuring the toolbar's buttons, click *Save* to store your changes.

Next, you will have to assign this recently-created *global* Programmable Toolbar to a user profile. To do so, follow the procedure below:

Step 12 Click on the *User* link and then the *User Groups* link.

Step 13 Select one of the existing user groups from the list, or click the *Add new* link to create a new user group. A window similar to the one shown below appears:

Step 14 In the User Profile section of the window, click on the *Toolbar 1* drop-down menu and select the desired global toolbar. All members of this user group will now have this toolbar appear on their Call Manager. You can repeat this step if multiple toolbars are desired and up to three global toolbars can be assigned to a user group.

Step 15 Click *Save* to store your changes.

10.4.10 Wireless Headset Hook Switch

A handset lifter is a device that can be used with phones that do not support the ability to place a call while the phone's handset is on-hook. The device is used in conjunction with a wireless headset worn by the user. When the user wishes to answer a call, she presses an activation button on the wireless headset which signals the handset-lifting device to manually lift the handset, thus generating dial tone and allowing the call to be placed. The small device attaches to the side of the phone and physically lifts the phone handset off-hook and keeps it there for the duration of the call.

As technology has advanced, many phone systems have eliminated the need for a physical handset lifter and have incorporated an electronic version of this device (i.e. electronic hook switch) into the phone design. Thus, the phone can be instructed to go off-hook electronically (as opposed to mechanically) in order to produce dial tone.

ShoreTel has incorporated the electronic handset-lifting functionality into various IP phone models (see below for a list of supported models). These ShoreTel IP phones work with the **Plantronics CS50** wireless headset. Users who have purchased this supported headset model can answer or end calls by pressing the activation button on their headset when they hear their phone ring. If they are too far from their phone to hear it ring, the headset will generate an audible cue to announce incoming calls.

User Groups
Edit User Group

[New](#) [Copy](#) [Save](#) [Delete](#) [Reset](#) [Help](#)

Edit this record [Refresh this page](#)

Name:

COS - Telephony: [Go to this Class of Service](#)

COS - Call Permissions: [Go to this Class of Service](#)

COS - Voice Mail: [Go to this Class of Service](#)

Send Caller ID as Caller's Emergency Service Identification (CESID).

Send DID as Caller's Emergency Service Identification (CESID).

Account Code Collection:

Show Call Manager users a list of account codes when dialing.

Outgoing Trunk Groups (Access Code):

- Analog Loop Start (9)
- ATT PRI Trk Grp (9)
- Digital Loop Start (9)
- Digital Wink Start (9)
- E911 trnk grp (9)
- HQ Test PRI TG (9)
- TEST PRI Trunk Group (9)
- TEST PRI2 Trunk Group (9)
- XO PRI trk grp (9)

Simplified Message Desk Interface Mode:

User Profile:

Toolbar 1: [Go to this Toolbar](#)

Toolbar 2: [Go to this Toolbar](#)

Toolbar 3: [Go to this Toolbar](#)

© 1998-2006 ShoreTel, Inc. All rights reserved.

Figure 10-31 Associating a global toolbar to a User Profile

Details:

- This feature is currently supported on the following ShoreTel IP phone models:
 - IP565g
 - IP560g
 - IP560 – Newer models support this feature while older IP560 models do not. (To see if your IP560 supports this feature, flip the phone and check the label with the model number. If the model number ends with a suffix of "-03" or higher, the phone will support this feature. If the suffix ends in "-01" or "-02" the feature is not supported.)
 - IP265
 - IP230
 - IP212k
- Using this feature will defeat auto on-hook and off-hook behaviors.
- The feature is only supported in conjunction with Plantronics CS50 headsets.

To configure this feature via ShoreWare Director:

- Step 1** Launch **ShoreWare Director** and enter the user ID and password. Then click the *Login* button.
- Step 2** Click on the *Administration* link to expand the list (if it has not already been expanded).
- Step 3** Click on the *Users* link and then click on the name of the user for whom you will be configuring this feature.
- Step 4** Click on the *Personal Options* tab to display a window similar to the one shown below:

Users
Edit User

New Copy Save Delete Reset

General Personal Options Distribution Lists Workgroups

User Name: Jane Smith
Extension: 888-55400

Call Control Options:
Current Call Stack Size: 8
Ring Type: Ring 4
Automatic Off-Hook Preference: Speakerphone Headset Wireless Headset

Figure 10-32 Configuring Wireless Headset Hook Switch for a user

- Step 5** Select the *Wireless Headset* radio button.
- Step 6** Click *Save* to store your changes.

To configure this feature on an IP phone via the telephone interface:

- Step 1** Press the *Options* button on the IP phone, followed by the associated password and the # key.
- Step 2** Scroll to option 2. *Auto Off-Hook*.
- Step 3** Press the *Edit* soft key to select this option.
- Step 4** Scroll to option 3. *Wireless Headset*.
- Step 5** Press the *OK* soft key to confirm.
- Step 6** Press the *Done* soft key to confirm.

10.4.11 System Directory Record

When you add a new user to the system, a user entry is automatically made to the system directory. You can edit a user's system directory record by going to the *System Directory Entry* edit page, shown in Figure 10-33.

You can edit some or all of the fields on the *System Directory Entry* edit page at your discretion. The *First Name*, *Last Name*, and *E-mail Address* fields are automatically entered when you add a new user. If you modify these fields on either page, they are automatically updated on the other page.

Figure 10-33 System Directory Edit Entry Page

10.4.12 Distribution Lists

A distribution list lets a user easily send a voice mail message to multiple users at one time. Each distribution list has a descriptive name and distribution list number associated with it.

Distribution lists are added to the system from Section 12.1 on page 273.

Users can be associated with the distribution lists from either the *Distribution Lists* options on the *Edit User* page (Figure 10-34) or the *System Distribution List* edit page (see Section 12.1 on page 273). When a user is associated with a distribution list, that user receives messages sent to that list.

If the user clicks the *Distribution Lists* tab on the *Edit User* page, the *System Distribution List* page appears.

The *Distribution List Memberships* box shows the distribution lists that are currently available. Clicking their check boxes adds or removes the user from the indicated list.

Users can be associated with more than one distribution list.

10.4.13 Workgroups

The *Workgroups* tab on the *Edit User* page lets you edit a user's workgroup membership, as shown in Figure 10-35. Users can belong to multiple workgroups; however, a user's login status is the same for all workgroups of which he is a member.

The *Workgroups* box shows the workgroup lists that are currently available. Clicking the check boxes adds or removes the user from the indicated list. Click the *Logged In* check box to activate membership.

Users
Edit User

New Copy Save Delete Reset Help

* modified

General Personal Options **Distribution Lists** Workgroups Refresh this page

User Name: Adam Ant
Extension: 4509

Distribution List Memberships:

- Executives
- Managers
- Staff

Figure 10-34 Edit User Page (Distribution Lists Tab)

Users
Edit User

New Copy Save Delete Reset Help

* modified

General Personal Options Distribution Lists **Workgroups** Refresh this page

User Name: Adam Ant
Extension: 4509

Make this user an agent of the following workgroup:

- Support Center
- Sales
- Special Promotions

Agent state: Logged out Logged in In wrap up

Figure 10-35 Edit User Page (Workgroups Tab)

10.5 Notify Users

The *Notify Users* page, shown in Figure 10-36, enables you to notify users that their ShoreTel client has been installed or upgraded. Once the user receives notification that the desktop client has been installed, the user can begin configuring personal options and use the client. To invoke the *Notify Users* page, click the *Notify Users* link in the navigation frame.

The options on the *Notify Users* page are as follows:

- All users: Select this option to notify all users throughout the ShoreTel 6.1 system.
- All users on this server: Select this option to notify all users on the local server.
- All users not yet notified: Select this option to notify all users who have not been notified.
- This one user: Select this option to notify this one user on the system.

Figure 10-36 Notify Users Page

- Send Email: Click this button to send the notification e-mail.

10.6 Anonymous Telephones

Anonymous telephones provide flexibility to the ShoreTel 6.1 system by making additional ports or IP phones available without assigning them to any particular user extension. When configured as anonymous telephones, these ports and IP phones cannot receive calls but do have access to dial tone. If they have the proper Class of Service (CoS) permissions, users can assign their extensions to these phones via the telephone user interface. Once the port or IP phone is assigned an extension, it will receive calls to that extension until the user unassigns it. For more information on how to use the Office Anywhere feature, see Appendix A, starting on page 391.

The *Anonymous Telephones* page (Figure 10-37) lets you configure anonymous telephone ports and IP phones. Click the *Anonymous Telephones* edit page from under the expanded *Users* link in the navigation frame.

Figure 10-37 Anonymous Telephones Edit Page

The *Anonymous Telephones* page first lists any *Vacated Telephones*. A vacated telephone is a telephone that is configured as the home port or IP phone of a user on the system, but that user is currently assigned to another telephone and no other user is assigned to the vacated home phone.

Click *Add This Record* to add a new anonymous telephone port. Click the *Delete* link next to a record to delete an anonymous telephone port from the ShoreWare system. This also disconnects any calls that are in progress on the port.

You can make multiple changes on the *Anonymous Telephones* page. You must click *Save* to save the changes.

- **Assigned User Group:** This lets you select a user group that you assign to an anonymous telephone port.
- **Jack #:** This is the name of the telephone jack associated with the vacated port. This is typically the physical telephone jack that the telephone plugs into. See 10.4.4 on page 192.
- **Switch:** This is the switch that the vacated telephone port is associated with. It can be either an analog port or an IP phone.
- **Port:** This is the physical switch port number or IP phone MAC address that identifies the vacated telephone.
- **Current User:** This is the name of the user currently using the anonymous telephone port.
- **Current Ext:** This is the extension of the user currently using the anonymous telephone port or IP phone.

10.7 Extension Lists

The *Extension Lists* list page is shown in Figure 10-38. Extension lists are lists of users that can be used by group paging and departmental auto-attendant. Any user in the system may be added. An extension list cannot be deleted if it is in use.

Administrative permission is required for a user to be allowed to configure extension lists.

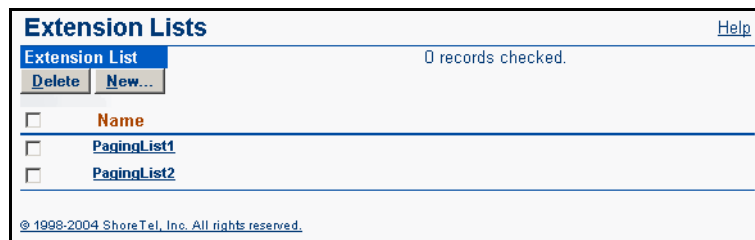


Figure 10-38 Extension Lists

Click an existing extension list to add users or click *New* to create a new list. The *Edit Extension Lists* page is shown in Figure 10-39.

Existing users are displayed and are available for selection. Highlight a user who is to be added to the extension list and click *Add*. The user moves to the Distribution List Members area of the page. A list member may be removed by highlighting a user in the Distribution List Members list and click *Remove*.

You can also select a Paging Extension associated with overhead paging system at a site to include it in a Paging Group. See Section 9.5 on page 145.

Users in an extension list intended for Paging Groups must belong to a user group that has enabled the Allow Overhead and Group Paging option.

- **Name:** The extension list must have a unique name.
- **Filter Users By:** Filter Users By is used to determine the order in which users appear in the Choose Members list. Users may be displayed in alphabetical order either by First Name or Last Name. The third alternative is to display users by extension number.

Figure 10-39 Edit an Extension List

In the Sort By drop down, select First Name, Last Name, or Extension. For more precise sorting, enter a character or number, as appropriate, in the First Name, Last Name, or Extension data boxes.

For example, to see extension numbers beginning with “3”, enter “3” in the Extension field.

To see all last names beginning with “Ban” enter those letters in the Last Name field.

To see all Smiths whose first names begin with “A”, enter “Smith” in the Last Name field and “A” in the First Name field.

- **Choose Members:** Click *Add* or *Remove* to move users into and out of an Extension List.

10.7.1 Paging Through a List

When a list becomes too long to display entirely on one page, you can page through the list by using the left, maximum left, right, and maximum right arrows next to the *Show Page* drop-down list. 50 items are displayed per page.

You can refine how users are displayed by using the *Sort By* drop-down list. Select from *First Name*, *Last Name*, or *Extension*. To go to a specific extension number or letter of the alphabet, use the *Filter Users By* field.

10.8 Batch Update Utility

Under *System Parameters*, the *Batch Update Utility* option helps you make changes to multiple users at the same time.

You can find a set of users based on any or all of the following criteria:

- Server
- User Group

- Personal Assistant
- Home Site
- Home Switch

You can globally change one of the following parameters for the selected users:

- Server
- User Group
- E-mail Domain Name
- Personal Assistant (for all call handling modes)
- Default Trunk Access Code
- Collaboration Server

To navigate to this utility, click the *Batch Update Utility* link in the navigation frame. You will be presented with the *Batch Update Utility* page (see Figure 10-40).

The screenshot shows the 'Batch Update Utility' page with a 'Status' tab. Below the tab is a section titled 'Filter Users based on the following criteria:'. There are five filter options, each with a checkbox and a dropdown menu:

- Assigned Server: ShoreDoc-HQ
- User Group: Make Selection
- Personal Assistant: [Text Field] Search
- Home Site: Make Selection
- Home Switch: Make Selection

At the bottom of the filter section are 'Find' and 'Reset' buttons.

Figure 10-40 Batch Update Utility Page

10.8.1 Batch Update

The Batch Update Utility allows you to find a set of users and globally change certain parameters. Due to the scope of this change, you are prompted to stop all ShoreWare voice services before running the batch update.

To use the batch update feature:

Step 1 Select one or more criteria to find the users.

This screenshot is similar to Figure 10-40, but the 'User Group' dropdown menu is open, showing a list of user groups:

- Make Selection
- Anonymous Telephones
- Executives
- House Telephones
- IP Telephones
- Managers
- Staff
- Voice Mail Notification

The 'Find' and 'Reset' buttons are still visible at the bottom.

Figure 10-41 Batch Update Utility—Find Criteria

Step 2 After you make your selection, click *Find*. You are presented with the results on the *Batch Update Utility Update Criteria* page.

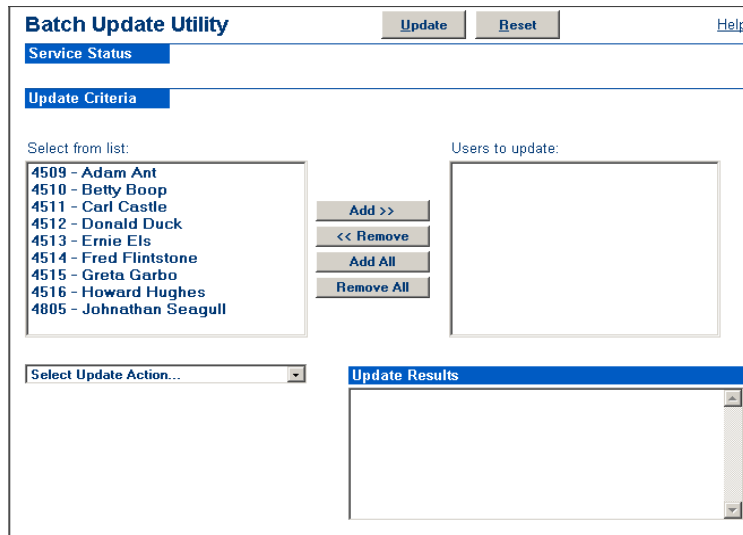


Figure 10-42 Batch Update Utility Update Criteria Page

Step 3 Select the users you want to change by adding them to the list on the right. You can add selected users from the left list, remove selected users from the right list, add all users (move all) from the left list, and remove all users from the right list.

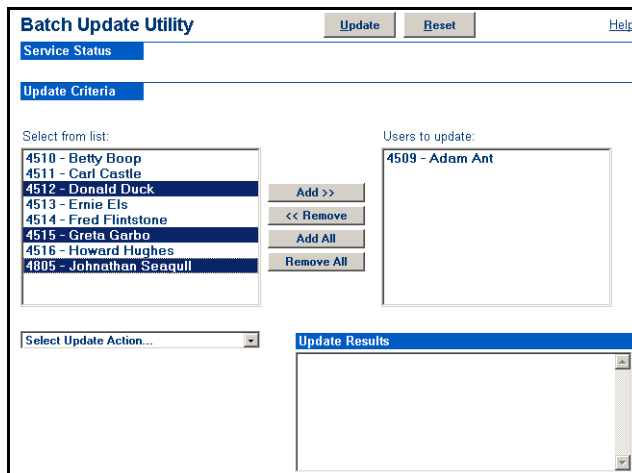


Figure 10-43 Batch Update Utility Page—Select Entities for Action

Step 4 Choose the parameter to globally change.

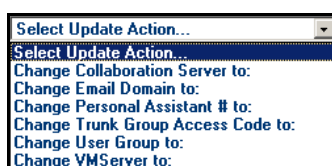


Figure 10-44 Batch Update Utility Page—Select Update Action

Step 5 Click *Update* to make the change on the system. Feedback on the action is presented in the *Update Results* field.

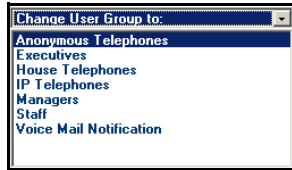


Figure 10-45 Batch Update Utility Page—Results List

When the change is complete, you can make additional batch updates.

Remember to restart the voice services when you have finished making your batch changes.

10.9 Call Handling Mode Defaults

Call Handling Mode Defaults are the set of call handling parameters assigned each time you add a new user. ShoreTel strongly recommends that you review and change these defaults before you add the bulk of your users.

Once a user is saved on the system, there is no relationship between the user's call handling modes and the default call handling modes. Changes to the default call handling modes do not affect the call handling modes of current users.

If you need to change the Personal Assistant of some or all users, you can use the Batch Update Utility discussed earlier in this chapter.

There are five default call handling modes, used for initializing each user's call handling modes. These modes provide a quick and easy way for users to change the way their inbound calls are handled.

The five default call handling modes are:

- Standard
- In a Meeting
- Out of Office
- Extended Absence
- Custom

Each of the call handling modes has the same configuration parameters.

Call handling modes specify how, when, and where calls are forwarded, and whether the user requires message notification when voice mail is received. The links in the *Edit Call Handling Modes* section under the *Personal Options* tab on the *Edit User* page bring up variants of the call handling mode pages. You can edit these copies for each user's personal options. Users can also modify these options from their desktop client applications.

Users can also change their call handling settings through a web interface on the ShoreTel server. For more information and the web interface URL, see the *ShoreTel 8 Planning and Installation Guide*.

10.9.1 Parameters

The *Call Handling Mode Default* parameters are as follows:

Standard Mode [Save](#) [Reset](#) [Help](#)
 Edit Default Values * modified

Edit this record [Refresh this page](#)

Call Forward Condition: Always No Answer/Busy Never

Always Destination: [Search](#)

Busy Destination: [Search](#)

No Answer Destination: [Search](#)

No Answer Number of Rings:

Personal Assistant: [Search](#)

Enable Find Me

Enable Message Notification

Schedule: [Edit this schedule](#)

Call Handling Note:

Figure 10-46 Call Handling Mode Default Values Edit Page

- **Call Forward Condition:** These buttons let you specify when calls are forwarded. The conditions are *Always*, *No Answer/Busy*, and *Never*.
 - The *Always* condition forwards calls to the number specified in the *Always Destination* parameter immediately when a call is received.
 - The *No Answer/Busy* condition forwards calls to the *No Answer Destination* after the specified number of rings, or to the *Busy Destination* immediately if the user's call stack is full.
 - The *Never* condition disables call forwarding.
 The recommended default is *No Answer/Busy*.
- **Always Destination:** When the *Always* call forward condition is selected, calls are forwarded immediately to this extension.
- **Busy Destination:** When the *No Answer/Busy* call forward condition is selected, calls are forwarded to this extension immediately if the user's call stack is full. The recommended default is *Voice Mail*.
- **No Answer Destination:** When the *No Answer/Busy* call forward condition is selected, calls are forwarded to this extension after the specified number of rings. The recommended default is *Voice Mail*.
- **No Answer Number of Rings:** When the *No Answer/Busy* call forward condition is selected, this parameter specifies how many times the phone rings before the call is forwarded to the *No Answer Destination*. The recommended default is three rings.
- **Personal Assistant:** Each user can specify a Personal Assistant, which is the destination to which a calling party is transferred upon dialing "0" in the user's mailbox. For example, executives often want callers transferred to their own executive assistant rather than to the operator when a caller dials "0" in their mailbox.

If no personal assistant is defined and a caller dials "0," the call is transferred to the site operator. If no site operator is defined, the call is transferred to the auto-attendant.

Users can also reach the Personal Assistant from the voice mail menu. By pressing 0 from the main voice mail menu, users can access the assistant. Alternatively, users can press 00 while listening to a voice mail message to reach the assistant. This can be helpful if a user is checking voice mails and wants to quickly reach the assistant to communicate something heard in a voice message.

The recommended default is an operator.

- **Enable Find Me:** This check box enables the *Find Me* feature by default for new users. When enabled, users can configure up to two numbers where they would like to receive calls that are forwarded from their voice mail. For more information, see Section 10.4.7 on page 205.
- **Enable Message Notification:** This check box enables message notification for this call handling mode. The manner in which the user is notified is determined by the user's message notification settings.

The recommended default is off.

- **Schedule:** This drop-down menu lets you select the schedule that will be associated with this call handling mode. For example, you may want to associate the Standard CHM with a schedule that is active from the hours of 9am to 5pm. Alternatively, you might wish to schedule the Out of Office CHM with a “graveyard” schedule that becomes active from the hours of 10pm to 6am.

For more information, see Chapter 14, starting on page 293.

- **Call Handling Note:** This text entry field lets you enter a note for each call handling mode. The call handling note is visible to users running the Operator Call Manager.

The recommended default is to leave this blank.

10.10 Escalation Profiles and Other Mailbox Options

The *Escalation Profiles and Other Mailbox Options* page can be accessed from a link on the *User's Personal Options* page. From here, you can configure Escalation Profiles to notify employees when a voice mail is received (which can be helpful in providing your customers with superior service/support after hours), and you can configure **Automatic Message Forwarding**, and **Email Delivery Options** for a user, so that users can be notified when their voice mailboxes are almost full.

The *Escalation Profiles and Other Mailbox Options* page is shown in Figure 10-47.

10.10.1 Parameters

The *Escalation Profiles and Other Mailbox Options* parameters are as follows:

10.10.1.1 EMAIL DELIVERY OPTIONS

- **Email Address:** This is the user's e-mail address. By default, it is automatically entered when you enter the user's first and last names in the First Name and Last Name fields. It consists of the first initial of the user's first name followed by the user's entire last name—for instance, **sdemont**. In addition, the @companyname.com domain is saved in a cookie on your workstation each time you save a user. This information is then presented as a default, which can be changed at your discretion.

Be sure to delete this field if the user does not have or use email.

Escalation Profiles and Other Mailbox Options
Jane Smith

[Edit this record](#) [Refresh this page](#)

Email Delivery Options:
 Email Address:
 Deliver Message as Email: Mark Message as Heard
 Send Email When Mailbox Is Full

Automatic Message Forwarding:
 Destination: None
 Mailbox:
 AMIS Mailbox:
 Delete Message after Forwarding

Message Notification Escalation Profiles:
 Escalation Notification Options: Escalate for Each New Message
 Escalate for First Unheard Message

Profile	Used By	Steps
1: Profile 1		1
2: <None>		0
3: <None>		0
4: <None>		0
5: <None>		0
6: <None>		0
7: <None>		0
8: <None>		0
9: <None>		0

Figure 10-47 Escalation Profiles and Other Mailbox Options Page

- **Deliver Message as Email:** Select one of three e-mail delivery options: *Disabled*, *Email Text Only*, or *Attach WAV File*. The *Email Text Only* option notifies the user of the time, duration, and Caller ID of the message that was recorded. The *Attach WAV File* option attaches the voice message to the email as a WAV file. Selecting the *Mark Message as Heard* results escalation profiles being disabled for messages that are delivered to "generic" email address (i.e. not the ones in the escalation steps).
- **Send Email When Mailbox is Full:** Select the *Send Email When Mailbox is Full* check box to enable Voice Mailbox Full Notifications. This feature sends users a notice informing them that their mailbox is almost full. This message is sent when the user's mailbox approaches maximum capacity and crosses a non-configurable threshold (i.e. space for only 10 messages remaining).

See Section 10.10.2 on page 238 for more information.

10.10.1.2 AUTOMATIC MESSAGE FORWARDING

- **Destination:** Messages may be auto-forwarded to a user, a workgroup, a route point, or a system distribution list. If the system distribution list includes AMIS destinations, they also receive the auto-forwarded message.

The default is None, meaning, no forwarding.

The destination may not be a broadcast distribution list.

- **Delete Message After Forwarding:** Select the checkbox to cause the automatic deletion of the message after forwarding. The default is not to delete.

10.10.1.3 MESSAGE NOTIFICATION ESCALATION PROFILES

- **Escalation Notification Options:** Escalation Paging is a traditional voice mail feature that allows support groups to offer round-the-clock service to their customers.

Customers can call into your system when they have a problem, and can leave a voice message. This causes the *Escalation Paging* feature to begin notifying (via email and/or pager/voice message) the appropriate personnel.

If the first person does not respond to the notification by listening to the customer's voice mail message within a certain time period, the next support person on the list is contacted, and so on, until as many as 10 people have been contacted.

You can choose to begin the notification process for each new message that arrives in the voice mailbox by selecting the *Escalate for Each New Message* radio button, or you can select the *Escalation Paging for First Unheard Message* radio button so that the notification process will activate for only the first unheard voice message (in which case subsequent unheard messages would be regarded as redundant and would not trigger another wave of notifications).

For more information on configuring this feature, see Section 10.10.3 on page 239.

- **Profile:** Name of the Escalation Notification Profile.
- **Used By:** This is a user's Call Handling Mode (CHM) that is associated with the Escalation Notification Profile. A profile can be associated with one or more CHM's
- **Steps:** This is the number of steps in that profile.

10.10.2 Configuring Voice Mailbox Full Notifications

The Voice Mailbox Full Notification feature offers a way for users to receive an alert that tells them their mailbox is almost full before it gets to the point that they stop receiving messages.

When a user's mailbox approaches its maximum capacity (and a non-configurable threshold has been crossed), the system sends users a notice informing them that their mailbox is almost full and that there is only enough room for 10 additional messages. Each time users log into voice mail, they will receive a notice telling them how much space remains. In this way, mailbox owners are given adequate notice that they must clean up their mailboxes and they are not caught off-guard by an unexpected (and unwanted) "mailbox full" notification.

Recall that the maximum number of messages a user can receive ranges from 0 to 500 and can be set on the "Class of Service - Voice Mail" window in Director. This flexibility implies that not all users in the system will have the same upward limit to the number of voice mail messages they can receive.

Details:

- The mailbox warning threshold occurs when there is room for only 10 more messages in a user's mailbox. This threshold is non-configurable and is the same for all users, regardless of total mailbox capacity.
- As a user's mailbox approaches its limit, a warning message will be played indicating that the user has room only for "n" number of messages where the value

"n" will be a countdown from 10 to 0. This message will be played when a user logs into the mailbox via the telephone user interface or PCM.

- The "almost full" notification will be played until a users delete their messages, thereby reducing the number below the threshold.
- When a mailbox has finally reached its limit, the mailbox owner will be notified (if this option has been enabled for this user) and a warning NT event will be logged.
- When a message is deleted, it is no longer counted against the total capacity for a user's mailbox.
- Deleted messages are temporarily held in a "deleted messages" folder. Up to 200 deleted messages can be temporarily held. Once this limit is reached, the mailbox will be considered full and the user will be unable to receive new messages until the deleted messages have been purged. If this happens, the mailbox owner will receive a notification telling him, "Your mailbox is full. No more messages will be accepted until you purge your deleted messages."
- Deleted messages can be manually purged by the user or automatically by the system. Automatic purging occurs on a nightly basis.

To enable the Voice Mailbox Full Notifications feature for a user:

- Step 1** From the *Escalation Profiles and Other Mailbox Options* page of a target user, verify that the Email Address field has a valid email address for this user.
- Step 2** Select the *Send Email When Mailbox is Full* check box.
- Step 3** Click *Save* to store your changes.
- Step 4** Repeat this process for each user for which you would like to configure mailbox full notifications.

10.10.3 Configuring Escalation Notification

The ShoreTel 6.1 system supports Escalation Notification. This voice mail feature allows your organization to know when your customers need help.

For example, if a customer in a small town calls his local utility provider to complain about a power outage at 4 a.m., it is possible nobody would be in the office at that hour to handle his call. However, with ShoreTel's Escalation Notification feature, the customer could leave a voice mail, and in doing so, he would set in motion a chain of events that would cause support personnel from the utility company to respond to his concerns.

The message left by the customer on the voice mail system would trigger the Escalation Notification feature to send out a page, phone call, or email to an employee in the support department of the utility company. If this first employee ignores the beeping pager, another person will be contacted, and so on. Each of those utility company employees specified in the escalation profile will be contacted until someone dials into the ShoreTel 6.1 system and listens to the customer's voice mail message and handles the problem. (See Figure 10-48 below.)

Details for creating an Escalation Notification profile:

- Each escalation profile has ten notification steps, allowing the system administrator to specify who will be contacted at each step and the method used to contact that person (i.e. phone call or pager). An email can be sent to that person in addition to the phone call or pager notification.

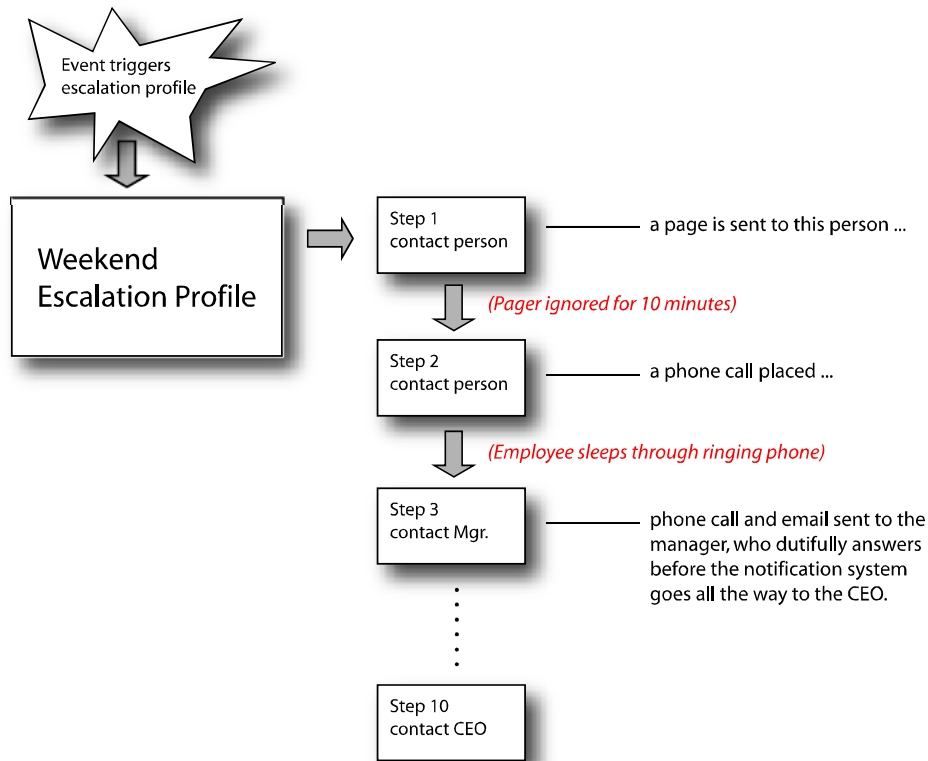


Figure 10-48 Anatomy of an escalation profile event

- A maximum of nine notification profiles are supported.
- Call handling modes can be associated with different notification profiles.
- If a message is left, and someone listens to it, the notifications will stop. However, if someone marks a message unheard, that will restart the notification process in the same way that receiving a new voice message will.
- Escalation notification is supported on all mailboxes, including user mailboxes (extension and mailbox users, mailbox-only users, SMDI mailbox-only users) as well as workgroup mailboxes.

To configure Escalation Notifications via ShoreWare Director:

Step 1 Launch ShoreWare Director and navigate to the *Escalation Profiles and Other Mailbox Options* link to display a window similar to the one shown in Figure 10-47.

Step 2 In the *Message Notification Escalation Options* section of the window, select the appropriate radio button. Options are:

- *Escalate for Each New Message* – Selecting this will cause a new wave of escalation notification messages to be sent out each time a new voice mail message arrives. If several messages arrive within a short period of time, those who are notified will receive multiple notifications (when perhaps one notification would have been all that was required).

- **Escalate for First Unheard Message** – Selecting this will cause the escalation notifications to begin at the receipt of the first voice mail message. Subsequent unheard voice mail messages will not trigger another wave of notifications as long as the first message remains unheard.

Step 3 Click on one of the links for the desired escalation profile (found at the bottom left of the *Escalation Profiles and Other Mailbox Options* page). The *Escalation Profile* page appears, as shown in Figure 10-49.

Escalation Profile
Jane Smith [Save](#) [Clear](#) [Reset](#) [Help](#)

[Edit this record](#) [Refresh this page](#) * modified

Name:

Repeat Count (0 - 200):

Escalation Steps 1 2 3 4 5 6 7 8 9 10

Timeout (0 - 3600): minutes

Urgent Only

Notification by Email:

Deliver Message as Email:

Email Address:

Notification by Phone:

Voice Mail Notification Method: Pager Phone None

Notification Number: Extension: External: (e.g. 9 +1 (408) 331-3300)

Pager ID:

Pager Data:

[Return to Escalation Profiles and Other Mailbox Options](#)

Figure 10-49 Editing the first step of this profile

Step 4 Enter the name of the profile in the *Name* field, if the profile has not already been named.

Step 5 Enter a value ranging from 0 to 200 in the *Repeat Count* field. This is the number of times the system will loop through the 10 steps of this profile before it quits trying to contact the various notification members. Selecting 0 will cause the escalation notification profile to execute once, without repeating. Selecting 1 will cause it to execute twice (i.e. once with one repeat loop).

Step 6 Enter the number of minutes in the *Timeout* field (0 - 3600) that should elapse before the next step within this profile is executed. (This is the amount of time a message recipient has to respond to the original voice mail before escalation occurs.)

Step 7 In the *Notification by Email* section of the window, click the *Deliver Message as Email* drop-down menu and select the desired option. (The default value is *disabled*, meaning that email notifications will not be sent.) Your choices are:

- **Email text only** – Select this option to have a text message sent to this user's email inbox. The message will contain basic information about the message (e.g. timestamp, sender, etc.)

- **Attach WAV file** – Select this option to have a copy of the voice mail sent to the designated user's email inbox. This will allow the recipient to play the message from his or her PC.

Step 8 Enter the address of the first person that should be notified in the *Email address* field.

Step 9 In the *Notification by Phone* section, select the *Pager* or *Phone* radio button to have a message sent to this user via one of these methods.

Step 10 Select the *Extension* or *External* radio button as appropriate, and enter the user's phone or pager number in the field.

Step 11 Click *Save* to store your changes.

Step 12 Click on the next *Escalation Step* (above the Timeout field) and repeat this process to configure up to ten steps within this escalation profile.

Any unconfigured steps will be skipped over when the escalation profile is executed.

Details:

- To “delete” an escalation profile, you can click the *Clear* button at the top of the escalation profile window. This will reset all of the fields for every step in this profile, although the profile is not actually deleted.

To create a mapping between the Escalation Notification Profile and a Call Handling Mode (CHM):

Step 1 With ShoreWare Director still open, click on the *Users* link and then click on the name of the desired user.

Step 2 Click on the *Personal Options* tab, and then click on the desired *Call Handling Mode* link (i.e. **Standard**, **In a Meeting**, **Out of Office**, **Extended Absence**, **Custom**) to associated an escalation profile with that mode. The CHM page appears, as shown in Figure 10-50.

Step 3 In the *Escalation Profile* drop-down menu, select the desired escalation notification profile.

Step 4 Click *Save* to store your changes.

Step 5 Repeat this process to associate different escalation notification profiles with each of the different Call Handling Modes, as needed.

Out of office Mode [Help](#)

Jane Smith

Edit this record [Refresh this page](#)

Call Forward Condition: Always No Answer/Busy Never

Always Destination: Extension: External: (e.g. 9+1 (408) 331-3300)

Busy Destination: Extension: External: (e.g. 9+1 (408) 331-3300)

No Answer Destination: Extension: External: (e.g. 9+1 (408) 331-3300)

No Answer Number of Rings:

Personal Assistant:

Enable Find Me

Enable Message Notification

Escalation Profile: [Edit this profile](#)

Schedule: [Edit this schedule](#)

Call Handling Note:

Figure 10-50 Selecting the escalation profile to associate with Out of Office CHM

Configuring Client Applications

This chapter describes ShoreWare applications that users access during their daily activities. Applications covered in chapter include:

- Call Manager
- Contact Center Integration with Call Manager
- Mobile Call Manager
- Conference Bridge

11.1 Call Manager

11.1.1 Introduction

Call Manager is the ShoreWare client application that manages an end user's personal calls, voice mail, and personal system settings. Five Call Manager types span the complete Call Manager feature set. Several Call Manager features require additional licenses or specific administrative authorization.

The five Call Manager types include:

- Personal Call Manager
- Professional Call Manager
- Workgroup Agent Call Manager
- Workgroup Supervisor Call Manager
- Operator Call Manager

Refer to the Call Manager User Manual for a complete list of features that each Call Manager type supports.

11.1.2 Installation

11.1.2.1 .NET Installation

Call Manager is a .NET Framework 2.0 application. The Call Manager installer automatically downloads the correct .NET version if it is not present during the upgrade or install. Users are then prompted to accept the End User License Agreement from Microsoft to proceed with the .NET installation. Internet connectivity is required during the installation process.

11.1.2.2 Communication Server Implementation

Instant Messaging is supported by ShoreWare Call Manager, allowing users to interface with third party Instant Messaging servers that support SIP/SIMPLE protocols. ShoreWare Instant Messaging is provided to Call Manager users through a Microsoft Office Communications Server (OCS) or through through a ShoreTel Converged Conference Bridge Server.

Microsoft OCS

The following are Microsoft OCS implementation guidelines as required for compatibility with ShoreWare 8.1.

- **OCS Versions:** ShoreWare is compatible with the following Microsoft OCS version:
 - OCS 2005
 - OCS 2007
- OCS software is available in three editions: Standard, Enterprise, and Enterprise Expanded. Standard Edition is typically sufficient for ShoreWare deployments with less than 5,000 users.
- **OCS system component requirements** include:
 - Windows Server 2003 or later
 - Windows Active Directory
- **License Requirements:** OCS requires one standard client access license (CAL) for each ShoreWare user enabled for Instant Messaging.
- **Server Configuration:** The OCS server must be configured for UDP transport. Call Manager does not support TLS transport.
- **Connectivity Testing:** To test ShoreWare server connectivity, log two OCS servers to the ShoreWare server and send an Instant Message between the OCS servers.
- **Additional Information:** Additional OCS information is available from the following sources:
 - the Office Communicator 2007 Resource Center at
<http://office.microsoft.com/en-us/help/HA102373941033.aspx>
 - “Microsoft OCS Requirements for ShoreWare Call Manager Instant Messaging Feature ST-10063”.

These application notes are posted on the support site and searchable through the Knowledge base.

ShoreTel Converged Conference Bridge Server

The ShoreTel Converged Conference Bridge Server Version 7.1 supports ShoreTel 8.1 instant messaging and presence information exchanges. Refer to the *ShoreTel Converged Conferencing Director Installation and Administration Guide* for more information.

11.1.3 Configuring Instant Messaging

Instant messaging is the real-time transmission of text between two system users. Instant messages are sent from and received through a message window. Instant messages do not use call cells. ShoreWare does not encrypt Instant Messages.

ShoreWare Instant Messaging complies with RFC 3261 – SIP: Session Initiated Protocol and RFC 3428 – Session Initiated Protocol (SIP) Extension for Instant Messaging. Instant Messaging requires ShoreWare access to Microsoft Office Communicator Server (OCS).

11.1.3.1 Setting up the System Server

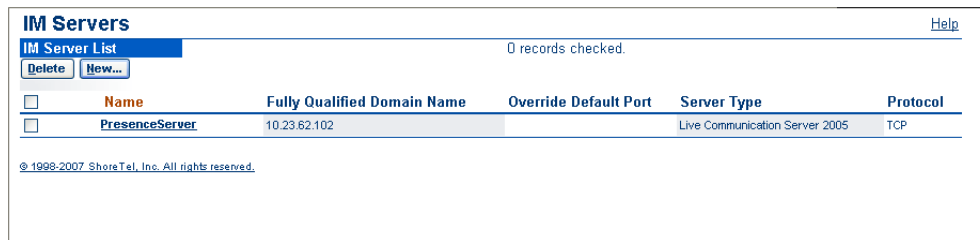
The following presence servers support ShoreWare Instant Messaging:

- Microsoft Office Communications Server, Version 2005
- Microsoft Office Communications Server, Version 2007
- ShoreTel Converged Conference Bridge Server Version 7.1

ShoreWare permits multiple presence servers on a system. Each user can be assigned to a maximum of one presence server. Users can exchange Instant Messages only with other users assigned to the same presence server.

To associate a presence server with the ShoreWare system:

Step 1 Open the IM Server list, as shown in Figure 11-1, by selecting *Administration -> IM Servers* from the Director main menu.

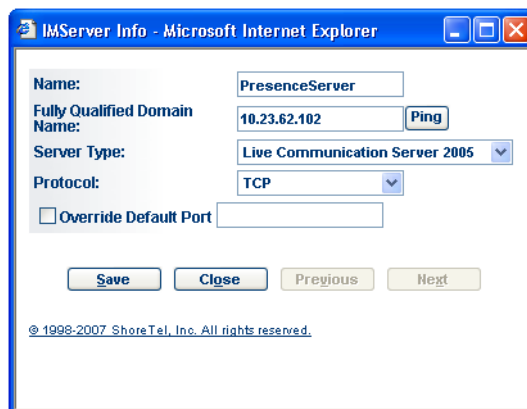


<input type="checkbox"/>	Name	Fully Qualified Domain Name	Override Default Port	Server Type	Protocol
<input type="checkbox"/>	PresenceServer	10.23.62.102		Live Communication Server 2005	TCP

© 1998-2007 ShoreTel, Inc. All rights reserved.

Figure 11-1 IM Server List

Step 2 Access the IM Server popup for a new server, as shown in Figure 11-2, by clicking the New button in the upper left corner of the IM Server List.



IMServer Info - Microsoft Internet Explorer

Name:

Fully Qualified Domain Name:

Server Type:

Protocol:

Override Default Port

© 1998-2007 ShoreTel, Inc. All rights reserved.

Figure 11-2 IM Server Popup

Step 3 Enter the information required to access the Presence Server

Step 4 Press the Save button in the IM Server Popup.

Step 5 Ping the IP address entered in the IMServer Info panel to verify that the address is accessible.

When changing the Presence Server version, access the IM Server popup and enter the presence version in the Server Type data entry field.

11.1.3.2 Configuring a User for Instant Messaging

The following Call Manager types can send and receive Instant Messages:

- Professional
- Workgroup Agent
- Workgroup Supervisor
- Operator

Configuring Instant Messaging for a User is a two step process:

Step 1 The administrator enables instant messaging for the user.

Section 11.1.3.3 on page 248 describes the process of enabling instant messaging for a user.

Step 2 The user configures instant messaging user options.

Refer to the Call Manager Users Manual for instructions on configuring instant messaging user options from the user's Call Manager window.

11.1.3.3 Enabling a Instant Messaging for a system user

To enable Instant Messaging for a system user:

Step 1 Open the Individual User list panel in Director by selecting *Administration -> Users -> Individual Users* from the main menu.

Step 2 Open the Edit User panel by clicking on the *First Name* of the desired user in the Individual User List.

Verify the First Name and Last Name parameters list the user for which you are enabling Instant Messaging.

Step 3 Verify that General is selected on the panel selection bar at the top of the panel.

Figure 11-3 displays the Edit User panel parameters that are relevant to configuring Instant Messaging.

Step 4 Verify that Client Type is set to either Professional, Workgroup Agent, Workgroup Supervisor, or Operator.

This parameter is located about halfway down the General panel.

Step 5 Select the desired Presence Server from the drop down menu for the Server parameter under IM Server.

This parameter is located at the bottom of the General panel.

Step 6 In the IM ID data entry field, enter the address through which the user will receive and send Instant Messages.

This address must be identical to the address configured for the user on the OCS server.

Step 7 Press the Save button at the top of the panel to enable the changes.

Figure 11-3 Instant Messaging Parameters on the Edit User panel

Step 8 Verify that the Presence Server is configured to authorize access by the address specified for the user in Step 6.

Consult the user documentation for the Presence Server for instructions on authorizing user access.

Step 9 Advise the User to verify access to the Presence Server by

Step a Opening the Options and Preferences panel from the User's Call Manager Main Window.

Step b Selecting Instant Messaging in the Menu on the left side of the panel.

Step c Entering the User's OCS User Name and Password in the Account Information section at the top of the panel.

Step d Pressing the Reconnect button.

The Status Bar in the Main window will display the result of the Call Manager attempt to access the Presence Server.

11.1.4 Presence

11.1.4.1 Description

Presence is a ShoreWare feature that identifies, uses, and distributes the availability of system users and other personal contacts. Presence information allows users to verify the availability of other users before attempting to contact them. Presence improves overall enterprise productivity by reducing calls to unavailable parties and by providing the enhanced ability to immediately schedule meetings, events, and communication sessions based on the availability of desired participants. Users can also choose when to receive Instant Messages by changing their presence settings

Call Manager define three presence settings for each user.

- **Telephony presence** indicates a user's availability to accept voice calls.

- **Instant Messaging presence** indicates a user's availability to engage in IM conversations.
- **Combined presence** is a single setting that indicates user availability on the basis of Telephony and IM presence status.

Call Manager automatically adjusts the presence status of users as they use system resources. Users can also manually adjust their status at any time.

A user can monitor the presence of a maximum of 500 other contacts. Administrators configure the number of contacts a user can monitor by setting the maximum contact list size, as described in Section 11.1.5.

11.1.4.2 Configuring a User for Presence

Telephony presence is available to users of all Call Manager types. IM presence is available to users authorized to use Professional, Operator, Workgroup Agent, or Workgroup Supervisor Call Manager. Director settings control user access to both presence types.

Configuring Presence for a User is a two step process:

Step 1 The administrator enables presence for the user.

Section 11.1.4.3 on page 250 describes the process of enabling presence for a user.

Step 2 The user configures presence user options.

Refer to the Call Manager Users Manual for instructions on configuring presence user options from the user's Call Manager window.

11.1.4.3 Enabling Presence for a User

To enable a user for telephony presence:

Step 1 Open the **Individual User List** in Director by selecting *Administration -> Users -> Individual Users* from the main menu.

Step 2 Open the **Edit User** panel by clicking on the *First Name* of the desired user in the Individual User List.

Verify the *First Name* and *Last Name* parameters list the user for which you are enabling Instant Messaging.

Step 3 Verify that **General** is selected on the panel selection bar at the top of the panel.

Step 4 Select the *Allow Telephony Presence* option, located in the middle of the panel.

Step 5 Click the **SAVE** button at the top of the panel to enable the changes.

To enable a user for IM presence:

Users that are authorized for Instant Messages are automatically configured for IM presence.

11.1.5 Setting the Maximum Contact List Size

Contacts are directory entries whose information is organized into specialized lists called Contact Lists to provide convenient access by the Call Manager user. Contacts are selected from directories accessible to Call Manager, including the System Directory, Personal

directory or Microsoft Outlook. Contact Lists, which are normally accessed from the Call Manager main panel, provide quick access to regular directory entries without navigating through the directory menus. Users can perform tasks on Contact List entries that are available from other directories such as initiating voice call or instant messages, handing active calls, and sending email or voice messages.

The maximum ShoreWare Contact List size is 500 contacts for each user. The size of each user's Contact List is controlled through a Class of Service setting.

To set the maximum Contact List size for a selected Class of Service:

- Step 1 Open the Class of Service list panel by selecting *Administration -> Users -> Class of Service* in the Director Menu.
- Step 2 Open the Edit Telephony Features Permissions panel by clicking the name of the selected Telephony Features Permissions class of service at the top of the Class of Service list panel.
- Step 3 Enter the maximum Contact List size in the *Max. Buddies Per User* field located at the top of the Edit Telephony Features Permissions panel, as shown in Figure 11-4.

Figure 11-4 Setting the Maximum Contact List Size for a Class of Service

The maximum Contact List size specified by a class of service is imposed upon a user when the class of service is assigned to that user.

11.1.6 Enable SoftPhone for a User

SoftPhone is a completely integrated software component of Call Manager. SoftPhone users participate in voice calls and listen to messages through a headset or speakers with a microphone without any additional hardware.

The **Edit User – General** panel in Director enables users to access to SoftPhone.

To enable SoftPhone access for a selected user:

- Step 1 Open the Individual User list panel in Director by selecting *Administration -> Users -> Individual Users* from the main menu.
- Step 2 Open the Edit User panel by clicking on the First Name of the desired user in the Individual User List.
Verify the First Name and Last Name parameters list the user for which you are enabling SoftPhone.

Step 3 Verify that General is selected on the panel selection bar at the top of the panel.

Step 4 Select the *Allow Use of SoftPhone* parameter, located in the center of the panel.

Step 5 Press the SAVE button at the top of the panel to enable the changes.

11.1.7 Video Calls

ShoreWare supports video calls between two Call Manager users that are communicating directly. Call Manager instances communicating through a ShoreWare server cannot use video. ShoreWare does not support server-based, star-configurations or interoperability with other video solutions. ShoreWare performs video calls through an SVC codec.

ShoreWare does not support video-only calls. Video is added to a voice calls as requested by the users. Video can either be added to an active voice call or established as a part of a new voice call.

IMPORTANT: *Each end user attempting video calls must verify that they are using the most recent driver and have installed all updates for their video card. Refer to the video card manufacturers for complete information.*

11.1.7.1 Supported Video Cameras

For a list of Video Cameras currently supported by Call Manager, refer to “Video Camera Requirements for ShoreWare Call Manager, Feature AN-10062”. These application notes are posted on the support site and searchable through the Knowledge base.

11.1.7.2 Enabling Video Calls for Users

The Edit User – General panel in Director enables users for video calls.

To enable Video Call access for a selected user:

Step 1 Open the Individual User list panel in Director by selecting *Administration -> Users -> Individual Users* from the main menu.

Step 2 Open the Edit User panel by clicking the First Name of the desired user in the Individual User List.

Verify the First Name and Last Name parameters list the user for which you are enabling Video.

Step 3 Verify that General is selected on the panel selection bar at the top of the panel.

Step 4 Select the desired option from the Allow Video Calls drop-down menu located in the center of the panel.

The drop-down menu options include

- *None:* The user cannot make video calls or add video to existing voice calls.
- *Standard Resolution:* The user can perform VGA resolution video calls.

Step 5 Press the Save button at the top of the panel to enable the changes.

11.1.7.3 Enabling Intersite Video Calls

The Video call site setting is a Class of Service parameter. Members of a class of service can only conduct video calls with other Call Manager users at the same site unless this parameter is set for intersite calls.

To enable intersite video calls for a class of service:

Step 1 Open the Class of Service list panel by selecting *Administration -> Users -> Class of Service* in the Director Menu.

Step 2 Open the Edit Telephony Features Permissions panel by clicking the name of the selected Telephony Features Permissions class of service at the top of the Class of Service list panel.

Step 3 Select the *Allow Intersite Video Calls* parameter on the top portion of the Edit Telephony Features Permissions panel.

11.2 Contact Center Integration with Call Manager

11.2.1 Description

ShoreTel integrates Contact Center Agent Toolbar functionality into Call Manager, allowing end users to control their contact center state and access Contact Center functions and Call Manager operations from a single, unified interface.

Figure 11-5 displays the non-integrated Contact Center Agent Toolbar. This interface allows Contact Center users to program buttons on the toolbar by assigning common functions and call center operations to the various buttons. Users can perform the desired operation with the click of a button.

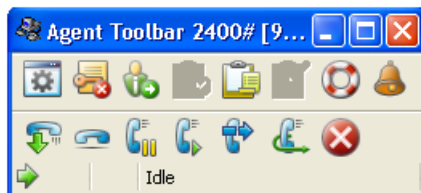


Figure 11-5 Contact Center Agent Toolbar

Contact Center functions are integrated into Call Manager by assigning Contact Center operation buttons into Programmable Toolbars, as described in Section 10.4.9 on page 217. Each toolbar can contain 24 buttons. To display additional buttons beyond those normally visible, click the arrow icon on right side of the toolbar. Figure 11-6 displays a Call Manager toolbar that contain Contact Center function buttons.

Details:

- Up to 24 buttons can be programmed in a single PCM toolbar.
- Toolbars can be created on a per-user basis or on a global basis. Toolbars created at the global level can be deployed to multiple users in the system.
- Three global toolbars maybe be deployed per User Group, with a maximum of 100 per system.

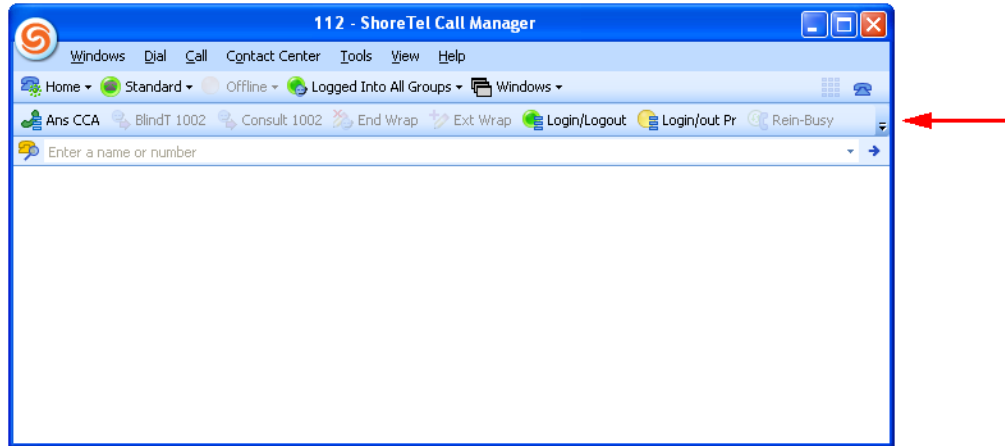


Figure 11-6 Call Manager toolbar containing Contact Center operation buttons

- Six personalized toolbars can be created for each user.
- A single user can employ 3 global toolbars, in addition to 6 personalized toolbars, for a total of 9 toolbars. With 24 buttons per toolbar, this allows a user to deploy a total of 216 programmable toolbar buttons. (24 buttons per toolbar X 9 toolbars = 216 buttons)

11.2.2 Accessing Contact Center Agent Toolbar Functions

The table below lists the various operations available in Contact Center and provides the new name for the operation in the integrated PCM (if applicable). Any operations that do not appear listed in the Contact Center Agent Toolbar column (in the middle of the table) do not have an analogous function in the integrated PCM. In many instances, this operation does not have an analogous operation in the integrated PCM because the Contact Center operation can be performed using one of the pre-existing functions available in the regular (i.e. non-integrated) PCM product.

Contact Center Operation Name	Operation Button or Path in Call Manager
Telephony Operations	
Answer	Answer Call Center Call
Set Callback - Reinsert Busy	Reinsert Busy Call
Set Callback - Reinsert No Answer	Reinsert Unanswered Call
Set Callback - Reinsert Terminate	Reinsert Terminated Call
ACD Operations	
Groups Manager	Select <i>Contact Center</i> -> <i>Agent Manager</i> from Call Manager
Login Primary Groups	Login Primary Groups
Logout from Primary Groups	Logout Primary Groups
Login Group	Login Group
Logout Group	Logout Group
End Wrap Up (Ready)	End Wrap-Up
Release with Code	Release With Code
Resume/ Release	Resume/Release

Table 11-1 Call Manager access methods to Contact Center operations

Contact Center Operation Name	Operation Button or Path in Call Manager
Supervisor Help	Supervisor Help
Transfer to Agent	Blind Transfer Agent
Wrap-Up Code	Wrap-Up Code
Wrap-Up Manual Control	Extend Wrap-Up
Accessing Applications Windows	
Desktop Wallboard	Select <i>Contact Center</i> -> <i>Desktop Wallboard</i> from Call Manager
Call Status	Select <i>Contact Center</i> -> <i>Call Status</i> from Call Manager
Queue Calls	Select <i>Contact Center</i> -> <i>Queue Calls</i> from Call Manager
Agent Log	Select <i>Contact Center</i> -> <i>Agent Log</i> from Call Manager
Contact Center Reports	Select <i>Contact Center</i> -> <i>Contact Center Reports</i> from Call Manager
Other Operations	
Pop-up window as client is launched	Set Agent ID
Execute an Application	Run Contact Center App. (or execute from Command Line)
Contact Center Functions Not Available in Call Manager	
Divert Incoming Call	
Manage List of Login / Logout ACD Groups	
Open Telephony Window	
Chat Tree	

Table 11-1 Call Manager access methods to Contact Center operations

11.2.3 Configuration

To configure integrated Call Manager – Contact Center for a user:

- Step 1** Launch ShoreWare Director and enter the user ID and password. Then click the Login button.
- Step 2** Click on the Administration link to expand the list (if it has not already been expanded).
- Step 3** Click on the Users link and then click on the Individual Users link.
- Step 4** Click on the name of the user whose toolbar you would like to configure.
- Step 5** Select *Enable Contact Center Integration*, as shown in Figure 11-7.

Step 6 Click Save to store your changes.

Next, you will need to configure the programmable buttons toolbar for this user.

- Step 1** Click on the Personal Options tab for the user who's system you are configuring, and then click the Program Personal Call Manager Toolbars link.
- Step 2** On the Program Personal Call Manager Toolbar window, click the New button to display a window similar to the following:
- Step 3** Enter a name for the toolbar and then click on the Function drop-down menu and select Contact Center.

Users
Edit User

New Copy Save Delete Reset

General Personal Options Distribution Lists Workgroups

First Name: Bonnie
Last Name: Hopkins
Number: 3644

Fax Support: User - Redirect

Client Type: Professional Enable Contact Center Integration

Allow Video Calls: None

IM Server:
Server: IMServer1
IM ID: bhopkins@anycorp.com

[Edit System Directory Record](#)

© 1998-2007 ShoreTel, Inc. All rights reserved.

Figure 11-7 Selecting the Contact Center client type

Program Call Manager Toolbar Buttons New Copy Save Delete Reset

Edit Toolbar Buttons Refresh this page

Name: Contact1

#	Function	Label	Target
1	Contact Center Blind Transfer Agent	Agent 1	Agent ID: 4123
2	Contact Center Login/Logout Group	Sales	Group Name: Sales_1
3	Contact Center Reinsert Busy Call		
4	Contact Center Answer Contact Center Call		
5	All Unused		
6	All Unused		

Figure 11-8 Assigning Contact Center operations to Call Manager toolbar buttons

Step 4 Then, select the desired operation from the drop-down menu. This operation will be associated with this specific toolbar button.

Step 5 Enter a label in Label field.

Step 6 If the function requires Target information, the required fields will appear to the right of the function. Enter the appropriate information as required for the type of operation the button will be performing. (For example, if the desired operation is Blind Transfer Agent, then an Agent ID field would appear in the Target area, allowing you to enter this information.)

If the target field for certain operations is left blank or is configured with invalid information, this will cause the operation to open a pop-up dialog box to collect more information (or valid information) when the user clicks on the associated button in PCM. This behavior applies to the following operations:

- Consult Transfer to Agent
- Blind Transfer to Agent
- Login/Login Group
- Logout Group
- Release with Reason Code
- Wrap-up with Code

Step 7 Continue programming the toolbar buttons as desired.

If a button is left unused or blank and this unused button is between other used buttons, PCM will interpret the sequence of blank buttons as a divider that will be visible on the toolbar.

Step 8 Once you have finished configuring the toolbar programmable buttons, click Save to store your changes.

Details

- Once you have finished the configurations described above, you can repeat this process of assigning operations to buttons for each new user that will be using the integrated PCM.
- When finished configuring buttons for each new user, follow the procedure below to configure the PCM clients.

11.2.4 Configuring Call Center Options through Call Manager

To configure the Contact Center options for a Call Manager user, refer to the Call Manager User's Manual.

11.3 Mobile Call Manager

Mobile Call Manager expands the remote user feature set by adding Call Manager functionality to mobile devices.

Mobile Call Manager comprises the following components:

- **Mobile Call Manager Server (MCMS)** is a ShoreWare server component that manages all communications with ShoreWare Mobile Call Manager clients. Installing the ShoreWare Server includes the automatic installation of MCMS.
- The **Mobile Call Manager (MCM)** client application is installed on each mobile device. Mobile Call Manager accesses ShoreWare functions, configuration information, voicemail, and calling history by communicating with the MCMS.

The MCMS is located on the main and distributed ShoreWare servers. Secure wireless communication sessions with specified mobile devices are conducted through a Reverse Proxy Server or a Blackberry Enterprise Server.

Mobile devices receive *MCM Client Installation Files* from the ShoreWare Server through a Blackberry Enterprise Server. After installing these files, the end user can configure the mobile device to access the ShoreWare server through an IP address provided by the system administrator.

ShoreWare Mobile Call Manager enables a mobile device to function as a ShoreWare extension and provides an interface, similar to Call Manager, for accessing ShoreWare client information. In addition to initiating calls from the Call History, Voice Mail, and

QuickDialer screens, end users can access their voice mail, configure call handling mode settings, set the active call handling mode, configure Office Anywhere settings, and activate Office Anywhere from the ShoreWare Mobile Call Manager user interface.

Mobile Call Manager supports up to 200 conventional users per ShoreTel system.

11.3.1 Supported Configurations

11.3.1.1 ShoreWare Mobile Call Manager Support – Languages

Mobile Call Manager software supports all elements of English (U.S.), English (U.K.), German, Spanish, Swedish, French, Dutch, Danish, Italian, and Norwegian.

Usability Enhancements

Mobile Call Manager now supports *smart type*. Smart type means that certain keys do not need a special key to enable character entry. On a BlackBerry, for example, a user can type a “.” or “-” without using the Alt key on numeric fields. This addition expedites the entry of IP addresses or extension numbers.

The left and right direction keys move the cursor so the user can insert one or more characters instead of being required to delete an entire field to correct an error. This feature is available on all devices except the older RAZRs (because they lack keyboard keys).

Localization Features

Localization support affects the following area of the product:

- **Western languages:** the supported languages are English (U.S.), English (U.K.), German, Spanish, Swedish, French, Dutch, Danish, Italian, and Norwegian.
- **Keyboard types:** on devices that have full keyboards, users can choose from programmed keyboard layouts for their location. For example, the top row of letters on American keyboards includes q w e r t y; German keyboards have q w e r t z; and French keyboards have a z e r t y.
- **Character set:** Mobile Call Manager utilizes the Latin-1 character set (see <http://htmlhelp.com/reference/charset/>). This character set includes all accented characters and other global symbols.
- **Ordering of time and date:** based on the language that a user specifies, Mobile Call Manager orders the date and time data according to the custom for that language. For example, with U.S. English, the order is mm/dd/yy; for U.K. English, dd/mm/yy; and for Dutch, dd-mm-yy. A table with the format of dates and times appears in Localized Date and Time Table section on page 268.
- **Accented characters:** Mobile Call Manager supports accented characters (diacritical marks). The keyboard for the selected language supports these marks, and the QuickDialer key filter also supports diacritical marks.
- The thumb wheel on the side of the device or other rotating character facility lets the user enter accented characters on the phone. The specific method for rotating through a character set depends on the model, yet the implantation is specific to Mobile Call Manager. (The ShoreTel model mirrors how a user enters characters on the device.)
- **Localized time on the phone:** The server uses a standard GMT format that the client converts to the time according to the time zone that is detected by the device. Therefore, the phone shows the time according to the time zone where the phone is located.

11.3.2 ShoreWare Mobile Call Manager Support – Devices

ShoreWare Mobile Call Manager is supported on the following devices:

- BlackBerry 8700
- Blackberry 7290
- BlackBerry 8800 Series
- BlackBerry 8300 Series
- BlackBerry 8100 Series
- Motorola RAZR V3
- Motorola RAZR V3xx Series
- Nokia E65 Series

This section describes the few functional differences based on the manufacturer or by model. Most models support all functions in Mobile Call Manager. This section lists the exceptions, so every model supports every function of Mobile Call Manager unless noted under a heading in this section.

11.3.2.1 Carrier Support

ShoreWare Mobile Call Manager is supported across carriers that allow installation of third-party applications onto the BlackBerry device.

The BlackBerry series 8100, 8300, 8700, and 8800 and the BlackBerry 7290 model are supported on any network that allows the installation of third party software. Although some mobile carriers restrict the downloading and installation of third-party software through their network, no known restrictions for the BlackBerry devices currently exist in any network.

Carrier support and restrictions on Motorola RAZR V3xx series and Nokia E65 series devices can be addressed by the end user's carrier.

11.3.2.2 Screen Transition

Screen transition refers to leaving one screen and going to another screen. Manufacturers offer menu-operations, buttons, or softkeys for moving to another screen. This section lists restrictions that manufacturers have for screen transitions by the clicking of a menu item, physical button, or soft key.

- Menu: The Nokia E65 and the RAZR do not have menu-activated screen transitions.
- E65, RAZR and RAZR V3xx screen transitions are done by softkeys.
- Button: None of the current models have button-only screen change activation.
- Soft key: Soft key is not supported by any of the BlackBerrys.

11.3.2.3 Personal Information Manager

Personal Information Manager (PIM) lets a device synchronize information from the Quickdialer feature in Personal Call Manager so that this information is automatically loaded and updated in Mobile Call Manager. PIM is supported by all models except the RAZR and RAZR V3xx.

11.3.2.4 Preview Voicemail

Voicemail cannot be previewed on the BB7290 because this model has no external speaker.

11.3.2.5 Font Size

Font size in the display cannot be changed on any of the BlackBerry models.

On the RAZR, RAZR V3xx, and Nokia E65, a user can change the font size in the Settings window.

11.3.3 Requirements

11.3.3.1 MCMS

The Mobile Call Manager Server (MCMS) is the ShoreWare server component that manages ShoreWare Mobile Call Manager client communications. The MCMS is installed on the ShoreWare HQ and DVM servers as part of the normal ShoreWare Server installation process. The MCMS contains no configuration parameters and requires no administrator intervention or monitoring during normal ShoreWare operations.

11.3.3.2 MCM Client Installation Files

MCM Client Installation Files are a set of files that provide the ShoreWare Mobile Call Manager client application to mobile devices. Clients download these files to their mobile devices from a server specified by the system administrator. *MCM Client Installation Files* are maintained and updated by the system administrator.

These files are originally placed on the ShoreWare Server as part of the ShoreWare installation, then moved by the system administrator to a server that can be accessed by the mobile devices of system end users.

11.3.3.3 Servers

Mobile devices must connect with the ShoreWare server to access MCMS services. Enterprise security requirements affect the structure of the network that provides application support. The following section describes servers that are typically used to implement ShoreWare Mobile Call Manager.

11.3.3.4 Blackberry Enterprise Server (BES)

The Blackberry Enterprise Server is a middleware application, offered by Research in Motion, that supports ShoreWare Mobile Call Manager by synchronizing email and PIM information through a secure back channel between the MCMS and BlackBerry mobile devices. The BES also supports Client Installation File downloads from ShoreWare to BlackBerry devices.

11.3.3.5 Reverse Proxy Server

A Reverse Proxy Server is a proxy server that is normally installed in front of web servers. Internet communications addressed to a web server are routed through the Reverse Proxy Server, which can process the request or pass the request to the specified web server. Implementing a Reverse Proxy Server within the corporate DMZ maintains the integrity of the corporate firewall and insulates the ShoreWare Server, along with other corporate assets, from public networks.

A Reverse Proxy Server provides secure wireless communication between Mobile Devices running ShoreWare Mobile Call Manager and ShoreWare server. A Reverse Proxy Server can serve the same role as a BES for conducting ShoreWare Mobile Call Manager wireless communication sessions.

Direct communication between an MCMS with the mobile devices is not recommended. This configuration compromises the integrity of the corporate firewall and exposes the ShoreWare Server and other connected corporate assets to attacks from public networks.

11.3.4 Administrator Configuration

ShoreWare Mobile Call Manager administration comprises two application activities: client application installation file delivery to the mobile devices and ShoreWare Mobile Call Manager communication sessions with the mobile devices. Configuring ShoreWare to support ShoreWare Mobile Call Manager requires the completion of the following tasks:

- Installing at least one additional server
- Acquiring the required ShoreWare Mobile Call Manager license
- Authorizing clients to use ShoreWare Mobile Call Manager

MCMS installation is performed with the installation of a ShoreWare server. The MCMS requires no additional installation or configuration. The following sections describe the tasks required to install and support ShoreWare Mobile Call Manager.

11.3.4.1 Server Installation (Headquarters and DVM)

Server configuration options:

- ShoreWare Mobile Call Manager communication sessions with a BlackBerry device are supported by a BES or a Reverse Proxy Server.
- Client Installation File downloads to a BlackBerry are supported through a BES.

Systems can use a BES server for all ShoreWare Mobile Call Manager activities.

BES

The BlackBerry Enterprise Server supports ShoreWare Mobile Call Manager communications between the ShoreWare server and BlackBerry devices. BlackBerry devices can also access *MCM Client Installation Files* located on the ShoreWare server through the BES. The BES implements a secure connection, similar to that of a VPN, to provide access for mobile devices to resources that are protected by the corporate firewall.

For information on installing a BES, refer to BES installation instructions provided by Research In Motion.

Reverse Proxy Server

Reverse Proxy Servers are used for conducting ShoreWare Mobile Call Manager communication sessions between the ShoreWare server and all supported devices. Reverse Proxy Servers cannot be used to deliver *MCM Client Installation Files* to mobile devices.

ShoreWare supports most available Reverse Proxy Server products. Appendix E in the Planning and Installation Guide describes the implementation procedure for an Apache Server installed on a Windows 2003 Server.

11.3.4.2 Client Installation Files

Installing the ShoreWare Server places the *MCM Client Installation Files* directly on the ShoreWare Server. BlackBerry devices can access the files directly from the ShoreWare Server if the network includes a BES.

Version numbering for Mobile Call Manager is in sync with the ShoreWare version number. The download format is xx.yy.zz. This format is required by the Nokia E65, RAZR, V3, and V3XX. In contrast, the BlackBerry does not have such a requirement. Nevertheless, an image with a number like 13.20.8700.0 is listed on the server as 13.20.87.00 for download activities, yet it appears in the device's About window as 13.20.8700.0.

Client Installation files can be delivered from a system that runs Windows 2003 Server and has IIS (version 5.0 or later). The IIS server must be configured to serve WML (Wireless Markup Language) files to the mobile devices. Table 11-2 lists the MIME types required to serve the installation files:

File Extension	MIME Type
.jad	text/vnd.sun.j2me.app-descriptor
.jar	application/java-archive
.wml	text/vnd.wap.wml
.cod	application/vnd.rim.cod

Table 11-2 Mime Types required to serve *MCM Installation Files*

To register the required MIME types:

- Step 1** Open the **Administrative Tools** panel by clicking the Windows *Start* button and selecting *Settings -> Control Panel -> Administrative Tools*.
- Step 2** Open the **Internet Information Services** panel by clicking on the *Internet Information Service* icon.
- Step 3** Open the **HTTP Headers** page, shown in Figure 11-9, by clicking the HTTP Headers tab at the top of the Internet Information Services panel.

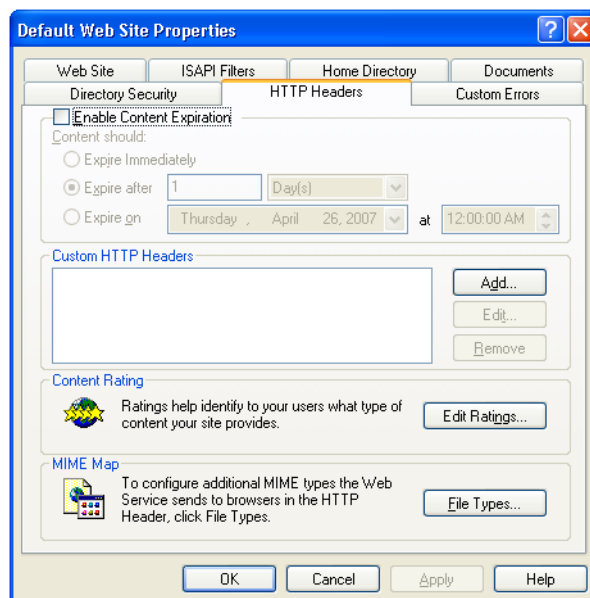


Figure 11-9 Internet Information Services panel – HTTP Headers page

- Step 4** Open the **MIME Types** panel by clicking the *MIME Types...* button in the bottom right corner of the HTTP Headers page.

Step 5 Open the MIME Types Dialog box by clicking the *New* button on the top right corner of the MIME Types panel, as shown in Figure 11-10.

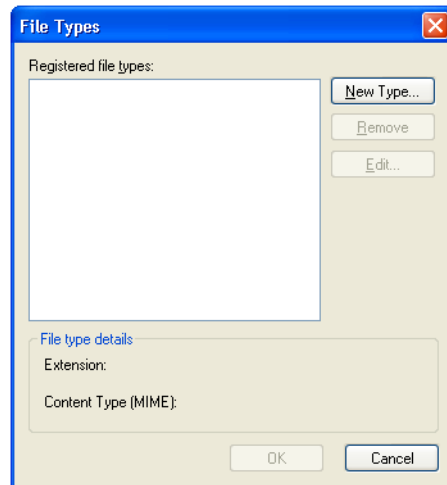


Figure 11-10 MIME Types panel

Step 6 Enter the Associated extension and content type data in the corresponding data entry field of the MIME Types Dialog box, shown in Figure 11-11, for one of the following types:

- Content type: `text/vnd.sun.j2me.app-descriptor`; extension: `.jad`
- Content type: `application/java-archive`; extension: `.jar`
- Content type: `text/vnd.wap.wml`; extension: `.wml`
- Content type: `application/vnd.rim.cod`; extension: `.cod`

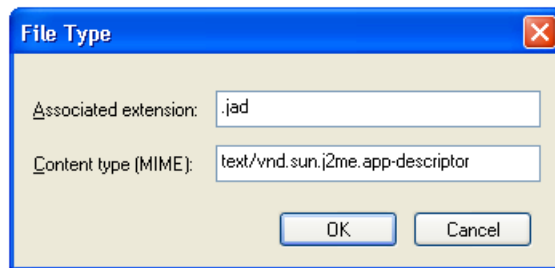


Figure 11-11 MIME Types dialog box

Step 7 Click the *OK* button to close the MIME Types dialog box and return to the MIMES type panel

The table lists the type entered in step 6.

Step 8 Repeat step 5 through step 7 for each type listed in step 6.

Step 9 Click *OK* buttons on each successive panels to save your changes and return to the desktop.

11.3.5 ShoreWare Director Configuration

11.3.5.1 Licenses

One ShoreWare Mobile Call Manager Keyed License is required for each client that is enabled for ShoreWare Mobile Call Manager. To view the number of ShoreWare Mobile Call Manager licenses available on your system, open the Director License Requirements panel, shown in Figure 11-12 by selecting *Administration -> System Parameters -> Licenses -> Requirements* from the Director menu.

Name	Required	Purchased
Keyed Licenses:		
ShoreWare System License (Enterprise Edition)	1	1
ShoreWare Additional Site License	3	8
ShoreWare Extension License	44	155
ShoreWare Mailbox License	43	155
ShoreWare SoftPhone License	4	25
ShoreWare Additional Language License	4	9
ShoreWare Mobile Call Manager License	1	1
ShoreWare Remote Web Reporting License	1	1
Self-audited Licenses:		
ShoreWare Personal Call Manager Software	1	0
ShoreWare Advanced Call Manager Software	34	20
ShoreWare Operator Call Manager Software	5	10
ShoreWare Workgroup Agent Call Manager Software	3	5
ShoreWare Workgroup Supervisor Call Manager Software	3	0
ShoreWare Remote Server Software	-	-
ShoreWare SIP Trunk License	0	0
ShoreWare Contact Center Supervisor	0	0
ShoreWare Contact Center Agent	0	0
ShoreWare TAPI Application Server	0	0
ShoreWare Phone API License	0	0

Figure 11-12 License Requirements page

11.3.5.2 Client Authorization

ShoreWare administrators grant ShoreWare Mobile Call Manager access to clients through ShoreWare Director. To enable ShoreWare Mobile Call Manager access for a ShoreWare user:

Step 1 Access the **Individual User** list by selecting *Administration -> Users -> Individual Users* from the Director menu.

Step 2 Open the **Edit User** panel for the desired user by clicking the name of the user in the User List.

Step 3 Verify that **General** is selected in the Properties bar located at the top of the Edit User panel.

The **Allow Mobile Call Manager** parameter is located on the lower section of the panel, as shown in Figure 11-13.

Step 4 Select the **Allow Mobile Call Manager** check box.

Step 5 Verify that **(Voice Mail Password) Must Change on Next Login** is cleared, as shown in Figure 11-13.

Step 6 Set the **Prompt for Password** parameter to specify the PIN entry frequency for the user

- **First Time Only** requires the user to enter the PIN only during the user's initial ShoreWare Mobile Call Manager access.

- Always requires the user to enter the PIN on each ShoreWare Mobile Call Manager access.

The screenshot shows the 'Users' administration page for editing a user. The user's name is Coloned Angus, with extension 888-55388. The 'Allow Mobile Call Manager' checkbox is checked, and the 'Prompt for Password' is set to 'Always'. The 'Must Change On Next Login' checkbox is also checked. Red dashed lines highlight these settings and the 'Allow Mobile Call Manager' checkbox.

Figure 11-13 Configuring an End User for ShoreWare Mobile Call Manager

11.3.6 Client Configuration Information Delivery

The administrator must provide the following information to each client authorized to use ShoreWare Mobile Call Manager:

- URL of the server through which mobile devices download the Client Installation Files
- IP address and Port Number through which the mobile device accesses the MCMS
- the client's user extension
- the client's voicemail password

11.3.7 Configuring Information upon First-Time Use

This section describes the first-time configuration requirements of the device.

11.3.7.1 Languages

The first screen that a user sees upon first-time startup of Mobile Call Manager under ShoreWare 8.1 is the language selection screen (Figure 11-14). The supported languages are English (U.S.), English (U.K.), German, Spanish, Swedish, French, Dutch, Danish, Italian, and Norwegian.



Figure 11-14 Language Selection Screen

Step 1 To select a language, use the track ball, wheel (at the right side of the device), or other device-specific selector to highlight a language.

Step 2 Press the wheel or press Select to use the highlighted language.

On devices that have a full keyboard, the keyboard selection screen appears after language selection. For devices that do not have a full keyboard, users are taken to the provisioning window.

11.3.7.2 Keyboard Type

After selecting a language, users see the keyboard selection menu on devices that have a full keyboard as shown in Figure 11-15:

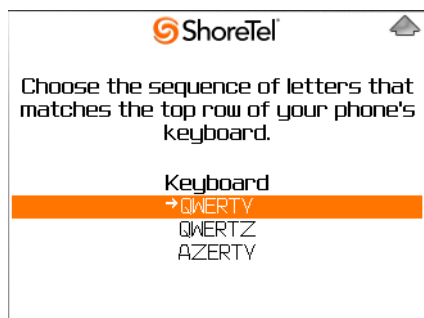


Figure 11-15 Keyboard Selection

Step 1 To select a keyboard, use the track ball or wheel at the right side of the device (or other, device-specific selector) to highlight the sequence of letters on the top row of the keyboard.

Step 2 Press the wheel or press Select to use the highlighted keyboard. After the user selects a keyboard, the informational screen appears in the selected language.

Step 3 Press Next to open the next screen, where specific user-details are entered. See next section, Provisioning User Details.

11.3.7.3 Provisioning User Details

The following user information is required for the first-time provisioning of a new phone:

- The IP address of the server that hosts the user account (not the DNS) is obtained from a system administrator. This server can be the BES or some other mechanism for reaching the server from the outside, such as a reverse proxy server.
- The Port number on the server that hosts the user account (obtained a system administrator).
- The user's extension.
- The password is the personal ID number (PIN): the user must enter this ID before hearing voicemail. Upon power up after the first-time specification of the password, the User Information screen does not show the password field.

After these four values are entered upon the initial run, Mobile Call Manager retains the configuration so the user does not need to re-enter it. An example of the regular User Information screen that is displayed after the initial configuration appears in Figure 11-16.



Figure 11-16 User Information Screen

The user can change the startup screen through the Settings window (“Default Start Page” in Figure 11-17).

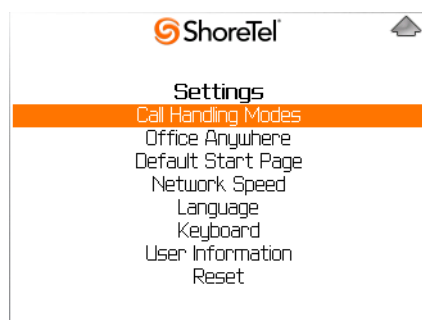


Figure 11-17 Settings Screen

After user information is initially entered, a status message momentarily appears, followed by a welcome message. After the welcome message, the Main Menu appears as the default startup window. The user can choose a different startup window by using the Settings window.

11.3.7.4 Specifying Accented Characters

The rotating character facility lets the user enter accented characters on the phone. This scheme follows the model used on the manufacturer's device. However, the implantation is specific to Mobile Call Manager, and this model mirrors the way a user enters characters on the device.

If a user enters the string "Andre" but actually wants "André Dupont," a wide variety of possibilities can come up (if they exist in the user's directory), for example:

Andre, Andrè, André, Andrê, Andrë, AndrÈ, AndrÉ, AndrÊ, AndrË

Now, the user can enter "André" to get André Dupont. Quickdialer can use the specific device's scrolling mechanism to select an accented character. Different devices have different mechanisms for scrolling through character choices. For the BlackBerry models, the user holds down a key and uses the trackball or thumb-wheel to scroll through the possibilities. Furthermore, all accent marks are available regardless of the selected language.

11.3.7.5 Localized Date and Time Table

This section consists of a table (Table 11-3) that lists the date and time that appears on a device based on the selected language.

Language	Date	Time
Deutsch	dd.mm.yy	08:48 23:59
Español	dd/mm/yy	8:48 23:59
Svenska	yy-mm-dd	08:48 23:59
Français	dd/mm/yy	08h48 23h59
Nederlands	dd-mm-yy	8:48 23:59
Dansk	dd-mm-yy	8:48 23:59
Italiano	dd/mm/yy	8:48 23:59
Norsk	dd.mm.yy	08h48 23h59
English U.K.	dd/mm/yy	08h48 23h59
English U.S.	mm/dd/yy	8:48 AM 11:59 PM

Table 11-3 Date and Time Formats by Language (Country)

11.3.8 Upgrading Mobile Call Manager

For an administrator, upgrading Mobile Call Manager requires only the installation of ShoreWare Version 8.1 to the server that runs Mobile Call Manager. This installation automatically places new versions of Mobile Call Manager's server components and client installer on the server.

Before upgrading a client device, the subscriber must delete the existing application from the device. To perform the upgrade, the subscriber accesses the URL that was used to download client software in ShoreWare Version 7.5. Access to the Mobile Call Manager is available through the following URL format:

`http://server/mcm/client/`

where *server* is either a valid DNS name or IP address of the Mobile Call Manager server. Upon entry of the correct URL, the device's browser is automatically directed to the latest file.

The server needs to be reachable from the mobile phone. This server could be a BlackBerry Enterprise Server (BES) or some other mechanism for reaching the server from the outside, such as a reverse proxy.

The following sequence illustrates an installation from a BES server:

1. The system administrator sends an email with a link to the location of the Mobile Call Manager software to the subscriber.
2. On the mobile device, the subscriber clicks on the email's enclosed link to get access to the download site.
3. The subscriber removes the current MCM version from the device.
4. The subscriber clicks the link leading to the download site on the browser window.
5. After the Mobile Call Manager software has been downloaded, the subscriber configures the client with the IP address and port allocated on the ShoreWare server for Mobile Call Manager.

As long as the device can reach the ShoreWare server, no other configuration or setup is required between the BES and the ShoreWare Server. (A company could restrict on access to its BES.)

Alternatively, the user can open a browser on the device and enter the URL of the server (in the format `http://server/mcm/client`). The server window opens with choices for locating the newest client software, as Figure 11-18 shows.

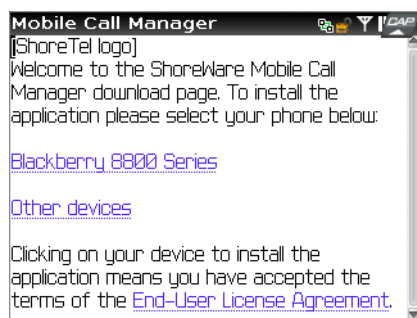


Figure 11-18 Accessing the Newest Client Software

The user selects the application for the device. In Figure 11-19, the software is for a BlackBerry 8800 series.

To download the client software, the user highlights and presses the Download button (Figure 11-20).

Upon successful software download a message (Figure 11-21) shows that the device is ready for the user to make choices for Mobile Call Manager's operation.

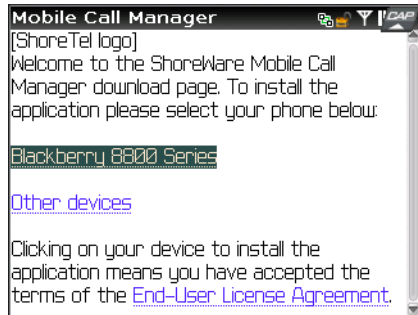


Figure 11-19 Choosing Specific Client Software

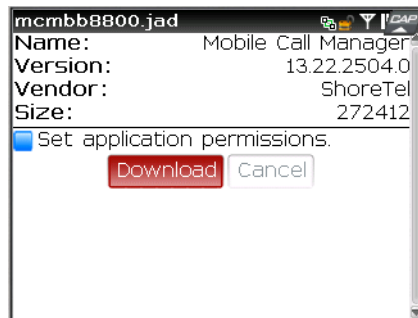


Figure 11-20 Starting the Download of the Selected Client Software

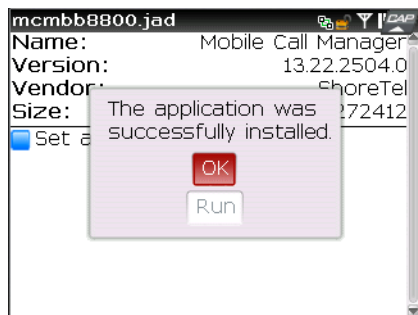


Figure 11-21 After Successful Download

11.4 Configuring Conference Bridge

This section explains how to set up conferencing servers to support Make Me conferencing and document sharing in a ShoreTel 6.1 system, providing information on:

- How to add a conference bridge, or collaboration server, to the system
- Other parameters impacting the use of conferencing

ShoreTel conferencing handles calls that share both voice and data. Up to 6 participants may share a conference and up to 20 Microsoft Word, Excel, or PowerPoint files. Make Me Conference calls and document sharing are independent of the ShoreTel Converged Conference Bridge.

Access to Converged Conferencing eDial version 5.6 through Personal Call Manager is not currently working. The feature, which can be accessed by launching PCM and going to *Options > Configure ShoreTel System > Login and Passwords > Converged Conference Login* relies on the SSL protocol and will be fixed in a later release.

11.4.1 Adding a Conference Bridge Server

The *Conference Bridges* link in the ShoreWare Director navigation frame provides access to the *Conference Bridges* list page as shown in Figure 11-22. To add a new conference bridge server, click *New*. To edit an existing server, click a server listed in the *Name* column. The *Conference Bridges* edit page appears as shown in Figure 11-22.

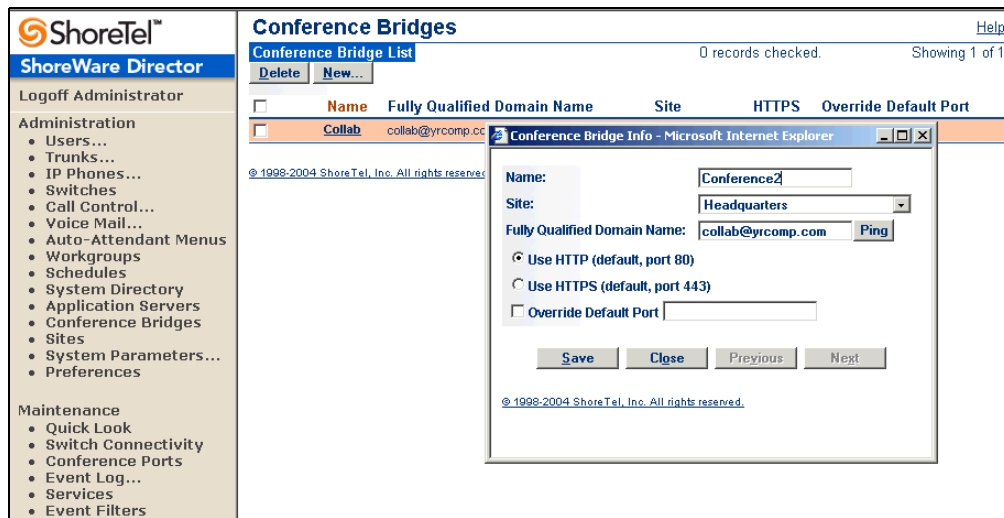


Figure 11-22 Conference Bridges Main and Edit Pages

Enter parameters for the new server as described below and then click *Save*.

11.4.1.1 Parameters

The parameters on the *Conference Bridges* edit page are as follows:

- **Name:** This is the name of a new or existing server.
- **Site:** This is the physical location of the server.
- **Fully Qualified Domain Name:** This is the server host name. If you enter the server host name, Director resolves the IP address for you when click *Ping this Server*.

You can also use the *Ping this Server* button to test the connectivity between your client PC and the new server.

- **Use HTTP (default port 80)**
Use HTTPS (default port 443)
Override Default Port with NNNN: Specify the communication port to be used to talk to the server.

For other parameters used in setting up conferencing, also see Section 10.4 on page 187. For individual users whose Class of Service permits document sharing, parameters need to be set to establish which conference server to use, a conference user ID, and a conference password. The class of service also specifies the maximum number of parties (3-6) that

may be in a single Make Me conference. Also see Chapter 9, starting on page 133 for information about the Make Me conference timeout parameter that is set in case a participant does not end the call correctly.

There may be up to six participants in a Make Me conference. Sufficient conference ports must be reserved on a ShoreTel switch to support the maximum simultaneous conferences you plan to support. For more information, refer to Chapter 5, starting on page 55.

11.4.1.2 Checking Conference Port Status

From the Maintenance link, you can check the status of conference ports. Go to the Switch Maintenance page and select *Conference Ports*.

ShoreTel™
ShoreWare Director
 Logoff Administrator

Administration

- Users...
- Trunks...
- IP Phones...
- Switches
- Call Control
- Voice Mail...
- Auto-Attendant...
- Workgroups
- System Directory
- Application Servers
- Conference Bridges
- Sites
- System Parameters...
- Preferences

Maintenance

- Quick Look
- Switch Connectivity
- Conference Ports
- Event Log...
- Services
- Event Filters

Documentation

- Administration Guide
- Planning and Installation Guide
- Telephone User Interface...
- Server...
- Client...

Conference Ports Last updated: 4/22/2004 9:23:20 AM (GMT -07:00) [refresh](#) [Help](#)
 Local time: 4/22/2004 9:24:13 AM (GMT -07:00)
[Go to Quick Look page](#)

Site	In-Use		Free		Percent
	Active Calls	Ports	Ports		
<u>Headquarters</u>	0	0	4	100%	
<u>El Dorado</u>	0	0	0	0%	
<u>Remote_Site</u>	0	0	0	0%	
Total	0	0	4	100%	

© 1998-2004 ShoreTel, Inc. All rights reserved.

Figure 11-23 Conference Ports List Page

Configuring Voice Mail

This chapter provides information about configuring the voice mail system, including adding and editing system distribution lists.

Click the *Voice Mail* link in the navigation frame to expand it to show the links for:

- System Distribution Lists
- Voice Mail Options
- AMIS Voice Mail

Also refer to Section 10.2.4 on page 183 for information about configuring your voice mail system.

12.1 System Distribution Lists

System distribution lists provide a mechanism for sending the same message to multiple users at one time. They are managed from the ShoreWare Director.

You add and edit system distribution lists from the *Voice Mail System Distribution Lists* page shown in Figure 12-1.

You can also add or remove a user from a system distribution list from the *Edit User* page (see Section 10.4 on page 187).

The ShoreTel 6.1 system allows users with the proper class of service to send a broadcast message to all mailboxes. Unlike system distribution lists, the broadcast distribution list cannot be edited. If necessary, you can selectively remove individual mailboxes from the broadcast list on the associated *Edit Users* page, Workgroup edit page, or Route Point edit page.

Description	Number
Broadcast	4600
Executives	4601
Managers	4602
Staff	4603

Figure 12-1 Voice Mail System Distribution Lists Page

The columns in the *System Distribution List* page are as follows:

- **Description:** This is the name of the system distribution list.

- **Number:** This is the number that is used for sending messages to members in the distribution list. Users can enter this number in either the ShoreWare client or when addressing a message from the telephone user interface.

To add or edit a system distribution list, click *Add new* or click the name of an existing list that appears in the *Voice Mail System Distribution Lists* table. When you click either of these items, the *Edit System Distribution List* page appears as shown in Figure 12-2.

Figure 12-2 System Distribution List Edit Page

12.1.1 Parameters

The parameters on the *System Distribution List* edit page (Figure 12-2) are described as follows:

- **Name:** This is the name of the distribution list.
- **Number:** This is the number that is used for sending messages to members in the distribution list. Users can enter this number in either the ShoreWare client or when addressing a message from the telephone user interface.

This number cannot go beyond the range of numbers that you defined in the *First System Distribution List Number* and *Last System Distribution List Number* fields in the section from Chapter 2, starting on page 15. Also, the first system distribution list number is reserved for future use.

- **Recorded Name:** The buttons that correspond to this item let you Record, Play, Erase, or Import a recorded name for the distribution list.

- Click *Record* to record a name for the distribution list.
 - Click *Play* to play back the recording.
 - Click *Erase* to erase the recording.
 - Click *Import* to import a sound file.
- **Language:** Select a language from the drop-down list.
 - **Select from List: Filter Users By**—If you want to search for users with certain extension numbers or names beginning with a specific letter, use the Filter Users By fields. You can sort by *Last Name*, *First Name*, or *Extension*.
If more names fit the criteria than can be displayed, use the *Show Page* controls. Either select a page number from the drop-down list or use the arrow keys to scroll through the list.
After selecting a name (or names) from the user list, click *Add* to add it (or them) to the *Distribution List Members* box.
If you want to remove a user from the directory list, select the user name from this box and click *Remove*.
 - **AMIS Systems:** This lets you add users on AMIS systems to the distribution list. To add an AMIS user to the distribution list, select the AMIS system where the user you want to add is located and click *Add*. A dialog box prompts you for the extension number of the user. Enter the number and click OK. The AMIS System ID and extension (Mailbox ID) appears in the distribution list box.

12.2 AMIS Voice Mail

The ShoreTel 6.1 system sends and receives voice mail messages to and from legacy voice mail systems using AMIS protocol Version 1 — Spec February 1992. To send voice mail messages to remote AMIS sites, ShoreTel 6.1 dials a phone number to access the remote system. Likewise, to receive voice messages from a remote system, the remote system must know the number to dial into the ShoreTel 6.1 system. To reach the ShoreTel 6.1 system, the remote system must be configured to dial a number that reaches an auto-attendant menu.

AMIS call support is enabled by default. Incoming AMIS voice mail is delivered in the same manner as other voice mail; however, replies cannot be sent. In order to send outbound AMIS voice mail, you must create AMIS systems in ShoreWare Director.

ShoreTel 6.1 negotiates the setup, handshaking, and teardown of AMIS system calls. Each voice mail requires a call over the AMIS delivery and call back numbers.

You can configure your AMIS systems for two addressing methods. If your system does not use off-system extensions, a *System ID* number is required to direct the voice mail to the correct site. When a user wants to send a voice mail to a recipient on an AMIS system, they must first enter the System ID followed by the mailbox number (extension).

Examples of Addressing with a System ID

System ID	Recipient Mailbox Number
8331	1234
8408331	45657

Table 12-1 Examples of Address with a System ID

If your system uses off-system extensions, these extensions become “off-system mailboxes.” In this case, users simply address the voice mail by mailbox number without needing to enter the System ID.

Before you create AMIS systems to remote sites:

- Enable AMIS messaging from the *Voice Mail Options* page (default is enabled).
- Set the permissions for the Voice Mail User Group to include dialing AMIS numbers. For more information, see Section 10.2.4 on page 183.
- Review the extension plans for all the systems to which you are connecting. Make sure they all use the same extension length and that there is no overlap of extensions.

After you have made these global settings, creating AMIS systems requires the following steps:

- Name the AMIS site and enter a System ID.
- Enter the phone number the ShoreTel 6.1 system calls to connect to each remote AMIS system.
- Enter the phone number that remote AMIS systems call to send AMIS messages. This number must reach an auto-attendant.
- If your system is using off-system extensions, select the extension range for each AMIS system.

12.2.1 AMIS Restrictions

Some restrictions are placed on AMIS voice messages. You should be aware of the following restrictions:

- ShoreTel 6.1 establishes a call to an AMIS system for each voice mail. If a voice mail is addressed to multiple recipients, ShoreTel 6.1 delivers as many as nine voice mails in a single call. If a voice mail has more than nine recipients, ShoreTel 6.1 makes additional calls until the voice mail is delivered to all recipients. You can optimize AMIS voice mail delivery by using distribution lists at the remote AMIS sites.
- The maximum message length permitted is eight minutes.
- After ten failed attempts to complete a call to an AMIS system, ShoreTel 6.1 disables the AMIS system and generates an event log.
- After ShoreTel 6.1 establishes an AMIS system call, it tries three times to complete message delivery to each recipient. If ShoreTel 6.1 fails to deliver a voice message after three attempts, it stops trying and returns the message to the sender. However, if the sender’s voice mailbox is full, they do not receive failed messages.
- Outbound voice mail messages for disabled AMIS systems are accepted and queued. To deliver queued messages, enable the AMIS system in question from *AMIS* edit page (see Figure 12-3).

12.2.2 Enabling AMIS

You enable AMIS systems from the *Voice Mail Options* edit page (see Figure 12-3).

12.3 Voice Mail Options

To reach the *Voice Mail Options* edit page shown in Figure 12-3, click the *Options* link in the navigation frame under *Voice Mail*.

Figure 12-3 Voice Mail Options Edit panel

The following set global AMIS parameters:

- **Minimum Message Length Accepted:** Specify minimum length a message must be to be acceptable. In milliseconds. The system default is 2000 milliseconds.
- **From Address for Email Notifications:** Specify the e-mail address to be placed in e-mail notifications about new messages.
- **Enable AMIS:** To enable AMIS systems support, click the check box (enabled by default). The *Enable AMIS* check box enables/disables all AMIS systems. Individual AMIS systems can be enabled and disabled from the *AMIS* edit page. For more information, see Section 12.3.1 on page 277.
- **Allow Incoming AMIS access to Broadcast DL:** To allow delivery of incoming AMIS messages to the Broadcast Distribution List, click the check box.
- **Allow Incoming AMIS access to System DL:** To allow delivery of incoming AMIS messages access to the System Distribution Lists, click the check box.

The Voice Mail application will automatically remove silence from voice messages. If the resultant message after silence removal is less than this minimum message length, the message is assumed to be a hang-up and will be deleted from the system.

12.3.1 Creating AMIS Systems

After you enable AMIS systems from the *Voice Mail Options* page, the next step is to create and configure the individual AMIS systems. Enter the AMIS delivery and call back number for each AMIS system you want to configure.

Expand the *Voice Mail* link in the navigation frame and click *AMIS*. The *AMIS Systems* list page appears as shown in Figure 12-4.

To add a new AMIS system, click *Add New*. To edit an existing system, click an entry in the *AMIS System* column. The *AMIS* edit page appears, as shown in Figure 12-5

The following section describes the *AMIS* edit page parameters.

AMIS Systems					
AMIS System	Enabled	System ID	Delivery Number	Mailbox Length	Call Back Number
Legacy Voice Mail	Yes	9555	+1 (408) 331-3300	4	+1 (408) 331-3300

Figure 12-4 AMIS Systems List panel

AMIS [Help](#)

[New](#) [Copy](#) [Save](#) [Delete](#) [Reset](#)

Edit AMIS

Edit this record [Refresh this page](#)

Name:

System Enabled

System ID:

Delivery Number: External (e.g. +1 (408) 331 3300)
 Private

Call Back Number: External (e.g. +1 (408) 331 3300)
 Private

Mailbox Length:

Off System Extensions: 3000 - 3999

Figure 12-5 AMIS Edit panel

- **Name:** In the *Name* text box, enter the name of this AMIS site.
- **System Enabled:** Click this check box to enable this AMIS system.

Outbound voice mail for this system is queued until the system is reset by clicking the *System Enabled* check box.

- **System ID:** The *System ID* defines the AMIS site where the voice mail for this system is delivered. The System ID plus a mailbox number identifies the site and the voice mail recipient. Plan the System ID to simplify the process of sending AMIS voice mail for your users. The System ID consists of an access code plus a site identifier.

The System ID must begin with a digit reserved for trunk access codes, although it can be different from other trunk access codes. To make the System ID intuitive to voice mail users, choose a site identifier related to the public numbers used at the site.

For example, if the voice mail delivery number is +1 (408) 555-1234, then System IDs like 8555 or 9408555 will be intuitive to your users. Generally, the shorter the System ID number, the easier it is to use.

The System ID plus the mailbox length cannot exceed 15 digits.

System IDs are required and can be single digits. Each AMIS system you create must have a unique System ID.

- **Delivery Number:** This is the number ShoreTel 6.1 calls to send AMIS voice messages to the remote system. An external number is a public PSTN number and a private number is an internal, off-system extension connecting to an intra-site PBX system.
- **Call Back Number:** This is the number on which you receive AMIS messages. An external number is a public PSTN number and a private number is an internal, off-system extension connecting to an intra-site PBX system.
- **Mailbox Length:** Set the mailbox length of the remote sites mailboxes (extensions). If you are using off-system extensions, the length must match the length of your extensions.

The System ID plus the mailbox length cannot exceed 12 digits.

- **Off-System Extensions:** If your system is using off-system extensions, select the extension range for each AMIS system. These extensions function as off-system mailboxes, allowing users to address voice mail to users on remote AMIS sites without entering a System ID. For more information, see Section 7.2.1 on page 96.

12.3.2 Disabling AMIS Systems

You can disable AMIS systems globally or by individual connections. Individual AMIS systems are automatically disabled when ShoreTel 6.1 fails to complete a call to an AMIS system.

To globally disable AMIS systems, unselect the *Enable AMIS* check box in the *Voice Mail Options* page (see Figure 12-3). To disable an individual AMIS system, from the *AMIS System* list page, double-click the name of the AMIS system you want to disable. The *AMIS* edit page appears. See Figure 12-3. Clear the *System Enabled* check box and click *Save*.

When you disable AMIS, ShoreTel 6.1 will not send or receive AMIS voice messages.

Users can address outgoing voice messages while the system is disabled. Outbound messages are queued until the individual AMIS system is re-enabled. Attempts to deliver to a disabled AMIS system fail.

12.3.3 Setting Voice Mail User Group Permissions

You must set the permissions for the Voice Mail User Group to allow ShoreTel 6.1 calls to the AMIS system delivery numbers you have configured. For instructions on setting these permissions, see Section 10.3 on page 185.

12.3.4 AMIS Test Mailbox

ShoreTel 6.1 allows you to designate a mailbox that a remote AMIS system can use to test AMIS features. When you address a voice mail to the AMIS test mailbox, ShoreTel 6.1 automatically replies with the same message.

Voice Mail reports are used by administrators to monitor, administer, and manage system resources and user activity.

12.4 Voice Mail Reports

When an administrator requests a Voice Mail report, the system retrieves statistics from the mainserver and all distributed servers. ShoreWare provides two types of reports from this information:

- **Summary Report:** The Summary Report lists voice mail resource data for each application server on the ShoreWare network.
- **Server Report:** The User Report lists voice mail usage information for the specified application server and users assigned to the application server.

12.4.1 Summary Report

The Voice Mail Servers Maintenance Summary panel displays the Voice Mail Summary Report. To open the Summary panel, shown in Figure 12-6, select *Maintenance -> Voice Mail Servers* from the Director menu.

Voice Mail Servers						
Site	Voice Mail Server	Mailboxes	Messages	Space Used (MB)	Free Space (MB)	Last Successful Backup
Headquarters	Headquarters	5	11	8006	32951	NULL
Headquarters	HQ DVM	2	1	7126	33831	NULL
Headquarters	HQ VMB90V	0	0	540	1431	08/31/2007 - 10:26:22 AM
Remote Site	RM DVM	6	6	6596	17870	NULL
Remote Site	RM VMB50V	0	0	471	527	NULL

● 25 to 50% left ● 25% Left

© 1998-2007 ShoreTel, Inc. All rights reserved.

Figure 12-6 Voice Mail Statistics: Totals Report

The table lists all application servers configured on the network. Each row corresponds to one application server and each column lists a server property or resource statistic.

The Summary Report includes the following fields:

- **Site:** This parameter lists the site to which the server is assigned, as specified by the Edit Application Servers panel.
- **Voice Mail Server:** This parameter lists the name of the server, as specified by the Edit Application Servers panel.
- **Mailboxes:** This parameter lists the number of mailboxes on the server.
- **Messages:** This parameter lists the number of messages that are stored in the server's mailboxes.
- **Space Used (MB):** This parameter lists the memory the server is using to store mailbox messages, user name recordings, auto attendant prompts, logs, and other data.
- **Free Space (MB):** This parameter lists the hard disk memory available on the server.

When less than 25% of memory capacity of the server is available, the panel displays a red icon next to the free space value for that server. When the available memory is between 25% and 50% of capacity, the panel displays a yellow icon next to the free space value.

- **Last Successful Backup:** This parameter lists the date that information on the server was backed up.

12.4.2 Server Report

The Voice Mail Servers Maintenance Server panel displays a voice mail statistics report for the specified application server. To open the Summary panel, shown in Figure 12-7, click the link of the desired application server on the Voice Mail Servers Maintenance Summary panel.

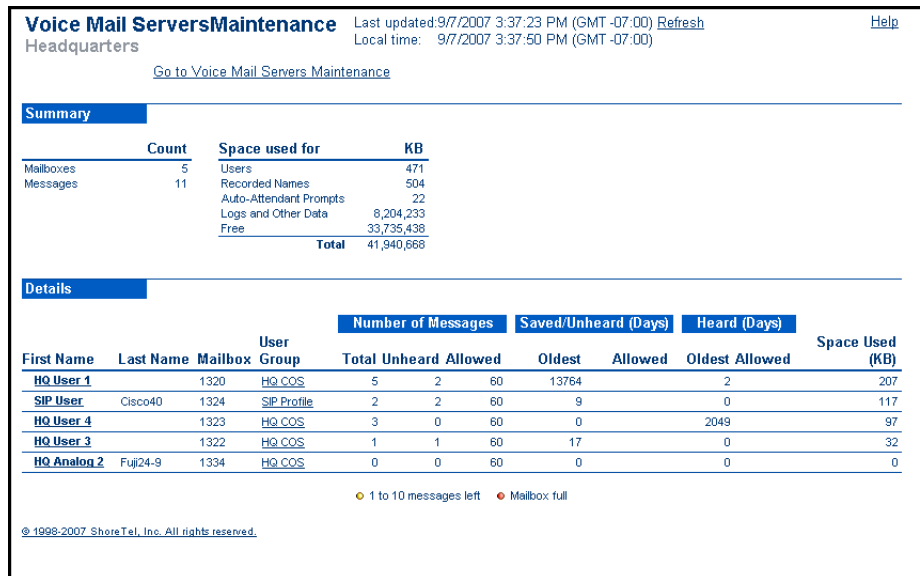


Figure 12-7 Voice Mail Statistics: User Report

The panel lists the name of the application server for which the panel is reporting statistics below the *Voice Mail Servers Maintenance* text in the upper left corner of the panel. The panel comprises two sections:

- **Summary:** The summary section lists resource usage and availability statistics for the specified application server.
- **Details:** The Details section lists resource usage statistics for users assigned to the application server.

12.4.2.1 Summary

The summary section, located on the top section of the Server panel, lists resource usage and availability statistics for the specified application server. Statistics listed in this section include:

- **Mailboxes:** This parameter lists the number of mailboxes on the server.
- **Messages:** This parameter lists the number of messages that are stored in the server's mailboxes.
- **Space Used for:** These parameters indicate the manner in which the server's memory is allocated:
 - **Users:** This parameter lists the memory used to store voice mail and mailbox configuration settings.

- *Recorded Names*: This parameter lists the memory used to store recordings of all user names.
- *Auto-Attendant Prompts*: This parameter lists the memory used to store auto-attendant prompts.
- *Logs and Other Data*: This parameter lists the memory used for log files generated by services hosted by the server.
- *Free*: This parameter lists the available unused memory resources.
- *Total*: This parameter lists the total memory resources on the server.

12.4.2.2 Details

The Details section lists resource usage statistics for the fifty largest mailboxes on the server. Mailboxes are sorted in order of the amount of disk space used to store their contents.

Each row corresponds to one mailbox on the application server. Each column lists a mailbox property or resource statistic.

The Details Report includes the following fields:

- **User Information**: These parameters identify the user to whom the mailbox is assigned. User information parameters are configured in the Edit User – General panel.
 - *First Name*: This parameter lists the first name of the mailbox owner.
 - *Last Name*: This parameter lists the last name of the mailbox owner.
 - *Mailbox*: This parameter lists the extension of the mailbox.
 - *User Group*: This parameter lists the User Group to which the mailbox owner is assigned.
- **Number of Messages**: These parameters list the mailbox capacity and contents statistics.
 - *Total*: This parameter lists the number of messages in the mailbox.
 - *Unheard*: This parameter lists the number of messages that are marked as unheard.
 - *Allowed*: This parameter lists the capacity of the mailbox. The number of messages that a mailbox can hold is configured through Class of Service settings.
- **Saved / Unheard (Days)**: These parameters list the age of the oldest message in the specified mailbox that has not been heard.
 - *Oldest*: This parameter lists the age, in days, of the oldest message that is marked unheard.
 - *Allowed*: This parameter lists when messages marked as unheard are removed from the server, in terms of the message age. This parameter is configured through Class of Service settings.
- **Heard (Days)**: These parameters list the age of the oldest message in the specified mailbox that has been heard.
 - *Oldest*: This parameter lists the age, in days, of the oldest message that is marked heard.
 - *Allowed*: This parameter lists when messages marked as heard are removed from the server, in terms of the message age. This parameter is configured through Class of Service settings.

- **Space Used:** This parameter lists the memory required to store contents of the specified mailbox. This statistic includes the memory required to store messages that are deleted but not purged.

Configuring the Auto-Attendant

This chapter provides information about configuring the auto-attendant. It includes the following sections:

- Multiple Auto-Attendants
- Menus
- Scheduling

The auto-attendant can answer incoming calls and transfer a caller to an extension, a mailbox, another menu, a workgroup, or a route point. It also includes a dial-by-name feature that transfers callers to the system directory, where they can connect to an extension by dialing the user's name.

13.1 Multiple Auto-Attendants

Multiple auto-attendants can be configured for different user groups or departments, and each auto-attendant configuration can have multiple levels of menu options.

There are no hard limits to the number of Auto-Attendants that can be configured in a ShoreTel 6.1 system. However, in most installations, the system can support up to 500 AA menus. This number may be affected by the complexity of your dialing plan.

When the "main" auto-attendant is reached, it provides options for forwarding calls to individual user extensions. It can also provide options for forwarding calls to the sales department and customer operations department auto-attendants. From the sales or customer operations auto-attendants, callers are given options that transfer calls to the appropriate extension.

The dial-by-name operation of the auto-attendant can be limited to a department or other organizational sub-group by associating the operation with an extension list. To create extension lists, see Section 10.7 on page 230. Only users that have been selected to be included in the dial-by-name list will be included. For more information, see Section 10.4 on page 187.

When callers are transferred back to the auto-attendant, either willingly or because of an error, they are returned to the default auto-attendant menu on the associated server.

13.2 Menus

The auto-attendant *Menus* page is where you begin the configuration to add a new auto-attendant menu or edit an existing menu.

13.2.1 Parameters

The parameters that appear on the *Menus* page are as follows:

The screenshot shows the ShoreTel ShoreWare Director web interface. The left navigation menu includes: Administration (Users..., Trunks..., IP Phones..., Switches, Call Control..., Voice Mail..., Auto-Attendant Menu, Workgroups, Schedules, System Directory, Application Servers, Conference Bridges, Sites, System Parameters..., Preferences). The main content area is titled 'Menus' and has a 'Help' link. Below the title is an 'Add new' link. A table lists existing menu configurations:

Name	Extension	Schedules		
		On-Hours	Holiday	Custom
Default	4700			
West Coast	4701			

At the bottom of the main content area, there is a copyright notice: © 1998-2004 ShoreTel, Inc. All rights reserved.

Figure 13-1 Auto-Attendant Menus List Page

- **Name:** This is the name of an existing auto-attendant menu configuration. Clicking an auto-attendant invokes the *Menus* edit page.
- **Extension:** This is the extension that is associated with an existing auto-attendant menu.
- **On-Hours:** This is the name of the *On-Hours* schedule, if any, that is associated with an existing auto-attendant menu.
- **Holiday:** This is the name of the *Holiday* schedule, if any, that is associated with an existing auto-attendant menu named in the *Name* column.
- **Custom:** This is the name of the *Custom* schedule, if any, that is associated with an existing auto-attendant menu.

13.2.2 Adding and Editing an Auto-Attendant Menu

To add a new or edit an existing auto-attendant menu, invoke the *Menus* page. From the *Menus* page, click *Add new*, or click an existing menu name in the *Name* column. This invokes the *Edit Menu* page, as shown in Figure 13-2.

13.2.2.1 Parameters

The parameters that appear on the *Menu* edit page are as follows:

- **Menu Name:** This is the name of the auto-attendant menu.
- **Extension:** This is the extension number associated with the auto-attendant menu. It must fall between the first and last menu numbers defined on the *Dialing Plan* edit page under *System Parameters*. See Section 2.2 on page 16.
- **DID:** When the check box is selected, a DID number is used to access the associated auto-attendant menu. If you are adding a new menu or editing an existing menu and you want to access it using a DID number, check this box and enter a DID number in the accompanying text-entry field. For more information about DID, refer to Section 7.2.3 on page 103.
- **DNIS:** Click this link to set up one or more DNIS mappings to this menu. For more information about DNIS, refer to Section 7.2.4 on page 103.

Menus
Edit Menu

Edit this record [Refresh this page](#)

Menu Name:

Extension:

DID: +1408331 (DID Range: +14083313916 - 3939)

DNIS:

Language:

Make Number Private

Allow Prompt Recording Using Telephone

Menu Password:

On-Hours
 Off-Hours
 Holiday
 Custom

Disable Monitor / Record Warning Tone.

Timeout (0 - 30000 milliseconds):

Prompt Text:

Prompt: no audio input

Schedule: [Go to this schedule](#)

Operation	Ext	Operation	Ext
0	Dial by first name	8	Dial by last name
1	<None>	9	<None>
2	<None>	*	<None>
3	<None>	#	<None>
4	<None>		Time out
5	<None>		Too many errors
6	<None>		Invalid entry
7	<None>		Multiple digits

Figure 13-2 Edit Menu Page

- **Language:** Select a language from the drop-down list. This is the language that will be used by the auto-attendant menu for responses such as “invalid entry”. Greetings must be recorded in this language.
- **Make Number Private:** Checking this box removes this number from the system directory and call handling destination lists.
- **Allow Prompt Recording Using Telephone:** Select this check box to enable the User Recording of Auto-Attendant Menus via the menu mailbox.

User recording of auto-attendant menus allows end users to dial into the system to record AA prompts in the same way that they would change their personal mailbox greeting (i.e. without having to access the recording interface through ShoreWare Director). This frees the system administrator from the task of recording AA menus, allowing him or her to delegate the task to more appropriate team members.

- **Menu Password:** A separate "Menu Mailbox" is created for each AA menu, allowing users to dial into the system to change the menu prompts. Each AA menu may have its own password and a unique, dialable number.

If a password is desired, enter the password in the field provided, and enter it a second time to confirm.

- **On-Hours / Off-Hours / Holiday / Custom:** These are the operating modes for a new or existing auto-attendant menu. You can configure them for different situations. Click the appropriate link to view the schedules. Schedules are set using the Schedules link. See Chapter 14, starting on page 293.
 - *On-Hours* mode lets you configure the auto-attendant to handle incoming calls during regular office hours.
 - *Off-Hours* mode covers all hours not scheduled in other modes. This is typically when the office is closed for the evening and weekend.
 - *Holiday* mode lets you configure how the auto-attendant functions on holidays.
 - *Custom* mode is used for single days that are not covered by the other modes such as a company special event.
- **Disable Monitor/Recording Warning Tone:** This check box can be used to stop playing the warning tone for call recording and monitoring if the tone is turned on in the Call Control page. To see where the warning tone is first set, refer to Section 9.8 on page 152.

WARNING ShoreTel, Inc. does not warrant or represent that your use of call monitoring or recording features of the Software will be in compliance with local, state, federal or international laws that you may be subject to. ShoreTel, Inc. is not responsible for ensuring your compliance with all applicable laws.

Before disabling the warning tone, you may wish to consult with legal counsel regarding your intended use.

- **Timeout:** Set a timeout between 0-30000 milliseconds. This is the time the caller has to perform an action.
- **Prompt Text:** Before recording a prompt for a new or existing auto-attendant menu, enter the text for the prompt in this field. This also provides a convenient record of your prompt if you should ever need to re-record the prompt.

This is an optional parameter.

Prompts on the ShoreTel 6.1 system can be imported into the system using μ -law, WAV file format. If you would like your prompts to match the voice of the ShoreTel 6.1 system, please contact Worldly Voices at www.worldlyvoices.com and request that “Connie” record your prompts. Worldly Voices provides this service with a rapid turnaround time for a nominal fee.

- **Prompt:** The *Record*, *Play*, *Erase*, and *Import* buttons associated with this parameter are used to record your auto-attendant menu prompt.

Click *Record* to record the prompt; *Play* to play it back; *Erase* to erase the prompt; and *Import* to import a prerecorded prompt from a sound file.
- **Schedule:** This drop-down list shows the auto-attendant schedules: *On-Hours*, *Off-Hours*, *Holiday*, and *Custom*. Select a schedule from this list and click *Go to this schedule* to invoke the schedule so that you can configure a new or existing auto-attendant menu’s schedule.

See Chapter 14, starting on page 293, for information about setting up schedules.
- **Operation:** Each item in the *Operation* drop-down list lets you select the action that is associated with its dialpad number. This number is located to the left of each Operation drop-down list. When prompted by the auto-attendant, the caller is asked to enter this number.

- *Dial by first name* lets the caller spell the user's first name from the dialpad. The auto-attendant then transfers the caller to the user's extension. To limit the dial list to a department or other organizational sub-group, select an extension list from the *Ext* column.
- *Dial by last name* lets the caller spell the user's last name from the dialpad. The auto-attendant then transfers the caller to the user's extension.
To limit the dial list to a department or other organizational sub-group, select an extension list from the *Ext* column.
- *Go to extension* lets the user enter the extension he needs. This functions the same as a transfer but without a voice prompt.
- *Go to menu* transfers the caller directly to the user's mailbox without ringing the user's extension. This is also used to send the caller to another menu. You must select the destination from the extension (*Ext*) *pop-up dialog box*.
- *Hang up* lets the caller disconnect the call.
- *Repeat prompt* lets the user hear the prompt again.
- *Take a message* lets the caller leave a message by selecting a user's extension.
- *Take a message by first name* lets the caller leave a message by selecting a user's name from the menu.
- *Take a message by last name* lets the caller leave a message by selecting a user's name from the menu.
- *Transfer to extension* transfers the caller to the user's extension where he or she can speak with the user or leave a message if the user does not answer. You must select a destination from the extension *pop-up dialog box*.

Dial by last name is supported by default.

- **Ext:** This *pop-up dialog box* lets you select the destination that is associated with the *Go to Menu* or *Transfer to extension* operation.
Used this field to select an extension list to be used by the Dial by First Name and Dial by last Name operations.
- **Time Out:** This drop-down list lets you specify the action that the auto-attendant takes when the caller does not press a dialpad key in a system-defined period of time. Typically, the action is *Repeat Prompt*.
- **Too Many Errors:** This drop-down list lets you specify the action that the auto-attendant takes when the caller presses an invalid key too many times in a row. You might specify a user extension, such as the operator, for this. Typically, the action is *Hang Up*. If no action is specified, *Hang Up* is invoked by default.
- **Invalid Entry:** This drop-down list selects an action to take when a key is pressed that the auto-attendant does not recognize. Typically, the action is *Repeat Prompt*.
- **Multiple-Digit:** This drop-down list lets you select a multiple-digit action that the caller takes. The choices are *None*, *Transfer to Extension*, *Take a Message*, *Go to Extension*, and *Go to Menu*. The default is *None*.
 - *None* assigns no multiple-digit operation to the menu.
 - *Transfer to Extension* assigns a multiple-digit operation to the menu and prompts the caller to dial directly into a user's extension.
 - *Go to Menu* assigns a multiple-digit operation to the menu and prompts the caller to dial directly into a user's mailbox.

13.2.3 Configuring an Auto-Attendant Menu

To configure an auto-attendant menu from the *Menus* page:

- Step 1** Enter the name of the menu in the *Menu Name* field.
- Step 2** If this is a new menu, enter the menu's extension in the *Number* field. If you are editing an existing menu, enter a new extension in this field if necessary.
- Step 3** If the menu will be associated with a DID number, check the *DID* check box and enter the DID number in the *DID* text-entry field.
- Step 4** To associate the menu with a DNIS number, click *Edit DNIS Map* and set the map.
- Step 5** Make Number Private only if you want to remove the number from the system directory.
- Step 6** Select the **Allow Prompt Recording Using Telephone** check box to enable the User Recording of AA menus, and enter and confirm a password for the associated mailbox.
- Step 7** Click the auto-attendant mode—*On-Hours*, *Off-Hours*, *Holiday*, or *Custom*—that will be associated with the menu.
- Step 8** Set a Timeout.
- Step 9** Enter the text that you will use for recording the menu's prompt in the *Prompt Text* field. This is optional.
- Step 10** Click *Record* to record the prompt. To hear the prompt, click *Play*; to erase it, click *Erase*. If you are importing a prerecorded prompt from a WAV file, click *Import* and select the file from the appropriate directory.
- Step 11** Select a schedule for this menu from the *Schedule* drop-down list, and click *Go to this schedule*. To configure the schedule, refer to Chapter 14, starting on page 293, for instructions.
- Step 12** Select the action that the auto-attendant takes in response to each supported digit from the *Operation* drop-down list.
- Step 13** Assign an extension to the *Operation* (if applicable) from the extension (*Ext*) *pop-up dialog box*.
- Step 14** Select an action from the *Time out* drop-down list, and select an extension from its extension *pop-up dialog* (if applicable).

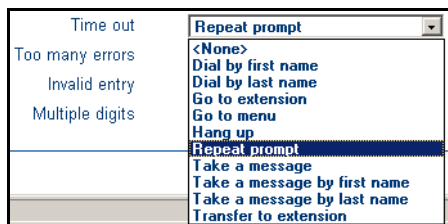


Figure 13-3 Timeout Errors Drop-Down List

Step 15 Select an action from the *Too many errors* drop-down list and select an extension from its extension *pop-up dialog box* (if applicable).

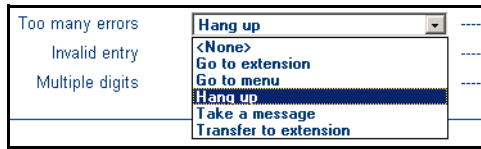


Figure 13-4 Too Many Errors Drop-Down List

Step 16 Select an action from the *Invalid Entry* drop-down list, and select an extension from its extension *pop-up dialog box* (if applicable). The options are the same as are available in the Timeout pop-up.

Step 17 Select an action from the *Multiple digits* drop-down list.



Figure 13-5 Multiple Digits drop-Down List

Step 18 Click *Save* to save the configuration.

13.2.3.1 Configuring Multiple Auto-Attendant Menus

Configuring multiple auto-attendant menus lets you add menus to your main menu so that callers can be directed to other departments in your company.

To add multiple auto-attendant menus to your main menu:

- Step 1** Go to the *Menus* page and configure the menu you are adding to the main auto-attendant menu, as described in the Section 13.2.2 on page 286.
- Step 2** Go back to the main auto-attendant menu's configuration on the *Menu* edit page.
- Step 3** Go to the *Operations* section on the *Menu* edit page and select *Go to menu*.
- Step 4** Associate the menu you are adding with a digit and the menu's extension. For example, if the menu's extension is 503, select this number from the *Ext.* drop-down list.
- Step 5** Click *Save* so that your changes are recorded.

You can define the auto-attendant's schedule when configuring an auto-attendant menu from the *Menus* page or by invoking the *Schedules* link. Refer to Chapter 14, starting on page 293 for information about establishing schedules.

The following schedule pages can be edited from the *Menus* page or from the *Schedules* link:

- On-Hours
- Holiday

- Custom

With the exception of the *Off-Hours* mode, each mode has a schedule configuration page. Off-hours is equal to all time not entered in the other schedules.

The following logic determines which schedule is active:

1. The auto-attendant first looks for the *Custom* schedule.
2. If the *Custom* schedule is not available, the auto-attendant looks for the *Holiday* schedule.
3. If the *Custom* or *Holiday* schedule is not available, the auto-attendant looks for the *On-Hours* schedule.
4. If the *Custom*, *Holiday*, or *On-Hours* schedule is not available, the auto-attendant looks for the *Off-Hours* schedule.

ShoreWare Director forms the Off-Hours schedule from all the hours not scheduled in the other modes. If you do not create a schedule for at least one of the other modes, the Off-Hours schedule will be all possible hours.

Configuring Schedules

14.1 Overview

This chapter provides information about configuring scheduling. Schedules can be used by Hunt Groups and by the Auto-Attendant.

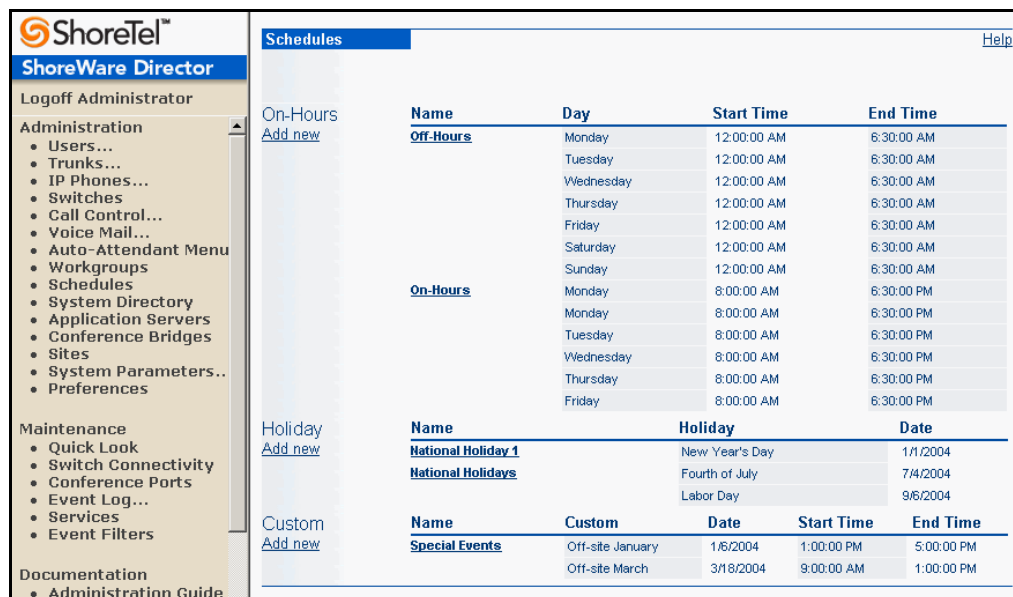


Figure 14-1 Schedules Main

The following schedule pages can be invoked from the *Schedules* page:

- On-Hours
- Holiday
- Custom

With the exception of the *Off-Hours* mode, each mode has a schedule configuration page. Off-hours is equal to all time not entered in the other schedules.

The following logic determines which schedule is active:

1. The auto-attendant or hunt group first looks for the *Custom* schedule.
2. If the *Custom* schedule is not available, the auto-attendant or hunt group looks for the *Holiday* schedule.
3. If the *Custom* or *Holiday* schedule is not available, the auto-attendant or hunt group looks for the *On-Hours* schedule.

Schedules					
On-Hours Add new	Name	Day	Start Time	End Time	
	Off-Hours	Monday	12:00:00 AM	6:30:00 AM	
		Tuesday	12:00:00 AM	6:30:00 AM	
		Wednesday	12:00:00 AM	6:30:00 AM	
		Thursday	12:00:00 AM	6:30:00 AM	
		Friday	12:00:00 AM	6:30:00 AM	
		Saturday	12:00:00 AM	6:30:00 AM	
		Sunday	12:00:00 AM	6:30:00 AM	
	On-Hours	Monday	8:00:00 AM	6:30:00 PM	
		Monday	8:00:00 AM	6:30:00 PM	
		Tuesday	8:00:00 AM	6:30:00 PM	
		Wednesday	8:00:00 AM	6:30:00 PM	
		Thursday	8:00:00 AM	6:30:00 PM	
		Friday	8:00:00 AM	6:30:00 PM	
Holiday Add new	Name	Holiday	Date		
	National Holiday 1	New Year's Day	1/1/2004		
	National Holidays	Fourth of July	7/4/2004		
		Labor Day	9/6/2004		
Custom Add new	Name	Custom	Date	Start Time	End Time
	Special Events	Off-site January	1/6/2004	1:00:00 PM	5:00:00 PM
		Off-site March	3/18/2004	9:00:00 AM	1:00:00 PM

Figure 14-2 Schedules List Page

4. If the *Custom*, *Holiday*, or *On-Hours* schedule is not available, the auto-attendant or hunt group looks for the *Off-Hours* schedule.

ShoreWare Director forms the Off-Hours schedule from all the hours not scheduled in the other modes. If you do not create a schedule for at least one of the other modes, the Off-Hours schedule will be all possible hours.

Descriptions of the pages and instructions for configuring these schedules follow.

14.2 On-Hours

The *On-Hours Schedule* edit page is shown in Figure 14-3.

14.2.1 Configuring the On-Hours Schedule

To configure the *On-Hours* schedule:

- Step 1 Click *Add new* or click an existing schedule from the *Name* column to view the schedule page.
- Step 2 Click a start time in a column, and hold down the left mouse button.
- Step 3 Drag the mouse pointer to the time that you want the action to stop, and release the mouse button.
- Step 4 Repeat Step 2 and Step 3 until you are finished scheduling the week.
- Step 5 You can modify time frames on the fly by clicking the right mouse button to invoke the options shown in Figure 14-4.
 - To use the same time frame throughout the week, click the *Fill Week* option.

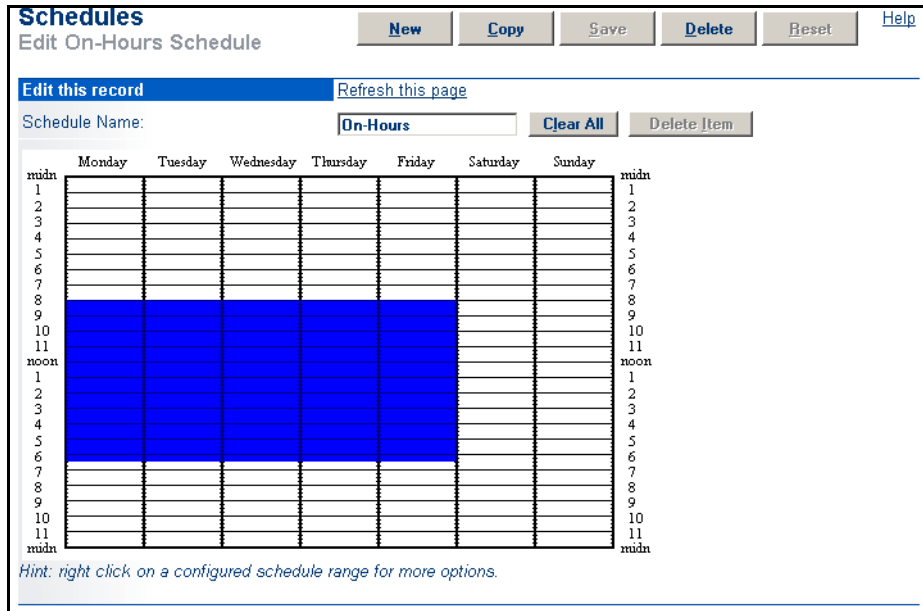


Figure 14-3 On-Hours Schedule Edit Page

- To edit an existing time frame, click *Edit*.
- To delete a time, select a time frame, click the right mouse button to bring up the options window and click *Delete*.

If you want to change the entire schedule, you must delete all entries in the schedule; otherwise, duplicate entries will be made.

Step 6 Click *Save* to save the schedule in the database.

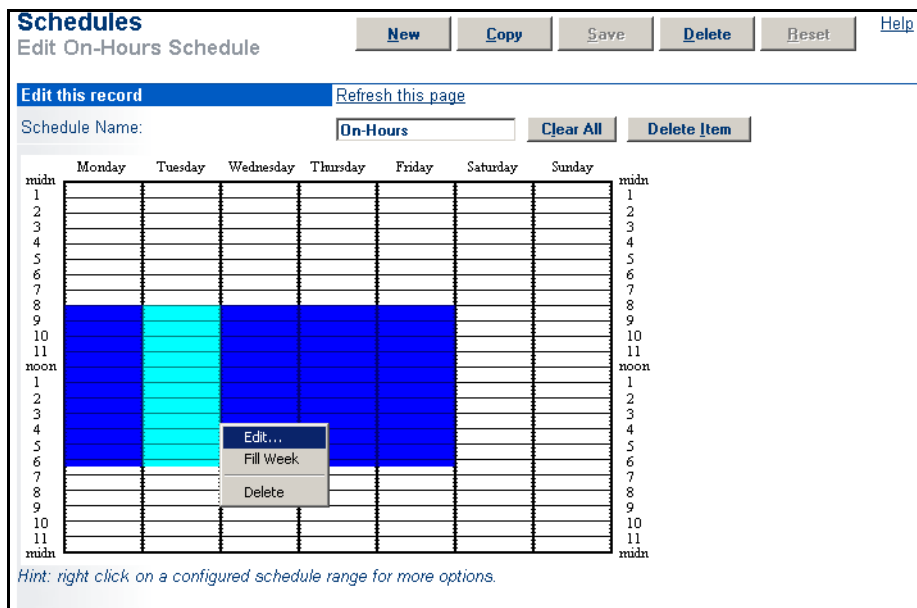


Figure 14-4 Time Frame Options

You can schedule a mode to start and stop multiple times in one day. For example, on Monday, you can set the schedule to start at 4:30 am and stop at 9:00 am, and then schedule it to resume at 2:00 pm until 5:30 pm by performing Step 2 and Step 3 again.

14.3 Holiday Schedule

The *Holiday Schedule* edit page is shown in Figure 14-5.

The screenshot shows the 'Schedules' interface for editing a 'National Holidays' schedule. At the top, there are buttons for 'New', 'Copy', 'Save', 'Delete', 'Reset', and 'Help'. Below these is a 'Refresh this page' link. The 'Schedule Name' field contains 'National Holidays'. Under the 'Holidays' section, there is an 'Add New' button and a table with two entries:

Holiday Name	Date	Delete Item
Fourth of July	7/4/2004	Delete Item
Labor Day	9/6/2004	Delete Item

At the bottom, it specifies the date format: MM/DD or MM/DD/YYYY.

Figure 14-5 Holiday Schedule Edit Page

14.3.1 Parameters

The parameters that appear on the *Holiday Schedule* edit page are as follows

- **Schedule Name:** Displays the name of a new or existing *Holiday* schedule. You can enter a name in this field for a new schedule or edit it for an existing *Holiday* schedule.
- **Holidays – Add New Item:** Click **Add New Item** to add a new holiday to the schedule; the *Holiday Name and Date* text-entry fields are added to the *Holiday Schedule* edit page.
- **Holiday Name, Date:** This lets you enter a name and date for a new *Holiday* schedule. The date format is *MM/DD* or *MM/DD/YY*.

Not entering a value for the year will repeat the same month and day throughout year.

This screenshot is identical to Figure 14-5, but the 'Add New' button under the 'Holidays' section is highlighted with a mouse cursor. The table now has a third empty row at the bottom, indicating that a new holiday entry is being added.

Figure 14-6 Holiday Schedule Edit Page (Add New Item)

14.3.2 Configuring the Holiday Schedule

To configure the *Holiday* schedule:

- Step 1 Select a holiday from the *Schedule Name* drop-down list or add a new one by clicking *Add New* and entering a name and date in the *Holiday Name* and *Date* text-entry fields.
- Step 2 Repeat Step 1 for all known holidays.
- Step 3 To delete a holiday from the schedule, click *Delete Item*.
- Step 4 Click *Save* to save the *Holiday* schedule to the database.

14.4 Custom Schedule

The *Custom Schedule* edit page is shown in Figure 14-7.

Custom Name	Date	Start Time	End Time	
Off-site January	1/6/2004	1:00:00 PM	5:00:00 PM	Delete Item
Off-site March	3/18/2004	9:00:00 AM	1:00:00 PM	Delete Item

Date format: MM/DD or MM/DD/YYYY
Time Format: HH:MM am or HH:MM pm

Figure 14-7 Custom Schedule Edit Page

14.4.1 Parameters

The parameters that appear on the *Custom Schedule* edit page are as follows:

- **Schedule Name:** This displays the name of a new or existing **Custom** schedule. You can enter a name in this field for a new schedule or edit it for an existing **Custom** schedule.
- **Custom Ranges – Add New:** Click *Add New* to add a new **Custom** schedule; the *Custom Name*, *Date*, *Start Time*, and *End Time* text-entry fields are added to the *Edit Custom Schedule* page.
- **Custom Name:** This displays the name of a **Custom** schedule. Enter the name of a new **Custom** schedule or edit the name of an existing schedule in this field.
- **Date:** This displays the date when a **Custom** schedule is used. Enter the date a new **Custom** schedule or edit the date of an existing schedule in this field. The format is *MM/DD* or *MM/DD/YYYY*.

Not entering a value for the year will repeat the same month and day every year.

- **Start Time:** This displays the start time of a **Custom** schedule. Enter the start time of a new **Custom** schedule or edit the start time of an existing schedule in this field. The format is *hh:mm:am* or *hh:mm:pm*.

- **End Time:** This displays the end time of a **Custom** schedule. Enter the end time of a new **Custom** schedule or edit the end time of an existing schedule in this field. The format is *hh:mm:am* or *hh:mm:pm*.

Schedules
Edit Custom Schedule

[New](#) [Copy](#) [Save](#) [Delete](#) [Reset](#) [Help](#)

[Edit this record](#) [Refresh this page](#)

Schedule Name:

Custom Ranges
[Add New](#)

Custom Name	Date	Start Time	End Time	
Off-site January	1/6/2004	1:00:00 PM	5:00:00 PM	Delete Item
Off-site March	3/18/2004	9:00:00 AM	1:00:00 PM	Delete Item
				Delete Item

Date format: MM/DD or MM/DD/YYYY
Time Format: HH:MM am or HH:MM pm

Figure 14-8 Custom Schedule Edit Page (Add New)

14.4.2 Configuring a Custom Schedule

To configure a *Custom* schedule:

- Step 1** Select a holiday from the *Schedule Name* drop-down list or add a new one by clicking *Add New* and entering a name and date for the new holiday in the *Custom Name* and *Date* text-entry fields.
- Step 2** Enter a start time in the *Start Time* field.
- Step 3** Enter an end time in the *End Time* field.
- Step 4** To delete a custom range, click *Delete Item*.
- Step 5** Click *Save* to save the *Custom* schedule to the database

Configuring Workgroups

This chapter provides information about configuring your ShoreTel 6.1 system for workgroups within your organization, such as Sales or Customer Care.

15.1 Overview

In the larger enterprise, there maybe small- to medium-sized groups working together as a contact center. The ShoreTel Contact Center Solution includes the ability for these workgroups to queue and distribute calls and provide agent and supervisor functions, as well as deliver reports about the workgroup activity.

15.1.1 Call Routing

The ShoreTel 6.1 system has many different call routing options to allow enterprises to configure their system and telephony services to match their needs. Callers reach the contact center by calling through a dedicated trunk pointed at the contact center, calling a DID or DNIS number directed to the workgroup, or calling and then navigating to an auto-attendant menu option linked to the workgroup.

Call handling options are combined with the system's schedules to provide even greater flexibility after callers reach the workgroup. Four call handling modes are configured for the workgroup that are scheduled for different times of the day. Each of the four call handling modes starts or stops according to the schedules and can invoke different options and transfer callers to different destinations when all agents are unavailable.

15.1.2 Call Distribution

The ShoreTel 6.1 system provides the contact center with flexibility for distributing callers to the available agents, as well as options for managing overflow. Inbound calls directed to the workgroup are managed by the ShoreTel server and distributed to agents in one of four administrator-configured patterns. When no agents are available, calls can be directed to a workgroup mailbox accessible by all agents or to a queue where calls can be held until an agent is available.

Distribution of the inbound calls is managed based on agent status. When agents are ready for calls, they log in and begin to receive calls. When they complete their day, they log out and calls are no longer delivered. In addition, the workgroup can optionally be configured so that all agents enter a “wrap-up” mode after every call. In “wrap-up” mode, agents remain logged in but do not receive new calls until the configured wrap-up time passes. This allows agents to complete any required updates to the customer records between calls.

The following is a summary of the call distribution and call overflow options:

Call Distribution Options

- Round Robin
- Top Down
- Longest Idle
- Simultaneous Ring

Call Overflow Options

- Hold in queue until an agent is available
- Transfer to another workgroup, extension, or external number
- Transfer to the shared workgroup mailbox

15.1.3 Call Queuing

The queue gives the administrator of the workgroup additional flexibility in managing his or her call flow. When all agents are busy, not logged in, or do not answer, callers can be directed to a queue where they are held until an agent is available to take their call. The queue offers up to five steps, each of which can be configured for different caller interactions and to enable caller-selected routing.

The following is a summary of the call queuing options:

Queue Step Configuration Options

- Announce the caller's estimated wait time
- Announce the configured prerecorded prompt
- Provide a menu to offer callers transfer options

Supported Menu Functions

- Customer inputs 0–9, *, #
- Transfers to menus, extensions, or mailboxes
- Repeat prompt or hang up the call

Call Queue Control Options

Each step can be skipped for call routing flexibility
Callers are on hold between steps for the configured “step time”
The last step repeats until the call is delivered

Other Features

Queue is an option only for on-hours call handling
Callers hear the main site's on-hold music while waiting

15.1.4 Workgroup Call Managers

The ShoreTel Call Manager applications for workgroup agents and supervisors provide the information the contact center representatives need in order to be effective, and also give them point-and-click control of their voice communications.

The Call Manager applications give agents and supervisors real-time call information, including available Caller ID, call duration, and call state. The call's detailed routing information is displayed so that agents know everyone the current caller spoke with in the enterprise before reaching the contact center. Additionally, the contact center's mailbox is displayed to every agent for accessing and handling the callers who chose to leave a message rather than wait for an agent.

Agents and supervisors have access to the real-time Queue Monitor. This application provides current information on what is happening in the contact center queue. It displays the number of callers, specific information about each, and how long people have been waiting.

The Agent Monitor is the supervisor's tool for managing the workgroup agents. It shows the supervisor the current login status of all the agents, whether they are on a call or not, and allows for changing the agent's status from the supervisor's position to manage contact center coverage.

The following is a summary of the workgroup Call Manager features:

Call Manager applications

- Display Caller ID, call duration, and call state.
- Display detailed routing information for calls.
- Display and access the shared contact center voice messages.
- Provide point-and-click access to the system's call handling features.
- Log in and log out of the workgroup call flow.

Real-time Queue Monitor

- Display a summary of the number of callers waiting and the longest wait time.
- Show a detailed view of the information about each waiting call.
- Display warnings when the number of calls or longest wait time exceeds the supervisor's thresholds.
- Display or control the call handling mode.

Supervisor's Agent Monitor

- Display the current login status of the agents in the workgroup.
- Show whether agents are on a call and how long they have been talking.
- Control agent's login status from the supervisor's position.

15.1.5 Intercom/Paging, Barge In, Record, and Monitor

Monitoring allows one party to eavesdrop on a call. It is a limited conference call where the monitoring party hears the other parties, but the monitored parties do not hear the monitoring party. Monitoring is undetectable by the parties being monitored, except by a warning tone. Monitoring is typically used in workgroups to evaluate agent performance.

A recording warning tone may be played to the customer during call recording and monitoring. The warning tone is enabled for the entire system using a Call Control option. For a specific call, it may be disabled by using an Auto-Attendant menu option. No tone is played during a Barge In call.

Barge In allows one party to join an existing call as a fully conferenced participant. When Barge In is initiated, a brief intrusion tone is played to the other participants and (if present) the monitoring warning tone is discontinued.

To simplify discussion of this feature, we will refer to three parties: the supervisor, the agent, and the customer. The supervisor initiates monitoring by selecting an agent. The agent is on a call with another party, the customer. The customer may be an external caller, but supervisors and agents must be on extensions.

In a monitored call, a supervisor hook flash is ignored. However, a hook flash by the other parties works the same as in a two-party call. In particular, an agent flash puts the call on hold and allows a consultative transfer or conference.

Because there is a limit of three parties in a conference call, if the agent or customer makes a consultative transfer or conference, the supervisor is automatically dropped. Similarly, if another party barges into a monitored extension, then the monitoring is dropped.

If a conference call is already in progress, it cannot be monitored. If monitoring is already in progress, no one else can monitor the call.

The supervisor can barge into a call he or she is monitoring. However it is not possible to revert a barge in to just a monitored call. If desired, the supervisor can hang up and restart monitoring.

After a barge in, the agent remains the controlling party of the call. A subsequent agent hook flash disconnects the supervisor, who was the last party added.

15.1.5.1 Intercom/Paging, Barge In, Record, and Monitor Telephony CoS Configuration

Each telephony class-of-service permissions has check boxes and radio buttons in ShoreWare Director for configuring Intercom/Paging, Barge In, Call Recording, and Call Monitoring. These options appear near the bottom of the window in Figure 15-1. For more information about setting permissions, refer to Section 10.2.1 on page 175.

Class of Service
Edit Telephony Features Permissions

[New](#) [Copy](#) [Save](#) [Delete](#) [Reset](#) [Help](#)

Edit this record [Refresh this page](#)

Name:

Max. Call Stack Depth:

Max. Parties in Make Me Conference:

Allow Call Pickup

Allow Trunk-to-Trunk Transfer

Allow Overhead and Group Paging

Allow Make Hunt Group Busy

Allow Extension Reassignment

Allow PSTN Failover

Show Caller ID Name and Number on Monitored Extensions

Allow Customization of IP Phone Buttons

Show Extensions with Different Prefixes in Directory

Allow Collaboration Features

Allow Recording of Own Calls

Directed Intercom / Paging: Allow Initiation
Accept: None All Only From: [Search](#)

Barge In: Allow Initiation
Accept: None All Only From: [Search](#)

Record Other's Calls: Allow Initiation
Accept: None All Only From: [Search](#)

Monitor Other's Calls: Allow Initiation
Accept: None All Only From: [Search](#)

Allow Call Handling Changes: Current Mode Detailed Settings

Allow External Call Forwarding and Find Me Destinations

Figure 15-1 Enhanced Class of Service Settings for Intercom, Barge, Monitoring

- **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 or extension specified in the associated field.
- **Allow initiation for barge in**—If this check box is selected, users within this COS may barge in on the calls of other system users. If cleared, then no barge in can be initiated.
- **Accept barge in**—Radio button choices are:
 - *Accept None*: If selected, users within this COS may not receive barge-in's from anyone.
 - *Accept All*: If selected, users within this COS may receive barge-in's from anyone else in this COS.
 - *Accept Only From*: If selected, users within this COS may only receive barge-in's from the person or extension specified in the field associated with this radio button.
- **Allow initiation for record others calls**—If this check box is selected, users within this COS may record the calls of other system users. If cleared, then no call recording can be initiated.
- **Accept record others calls**—Radio button choices are:
 - *Accept None*: If selected, users within this COS may not have their calls recorded from anyone.
 - *Accept All*: If selected, users within this COS may have their calls recorded from anyone else in this COS.
 - *Accept Only From*: If selected, users within this COS may only have their calls recorded by the person or extension specified in the field associated with this radio button.
- **Allow initiation for silent monitor**—If this check box is selected, users within this COS may monitor other system users. If cleared, then no monitoring can be initiated.
- **Accept silent monitor**—Radio button choices are:
 - *Accept None*: If selected, users within this COS cannot be monitored by anyone.
 - *Accept All*: If selected, users within this COS can be monitored by anyone else in this COS.
 - *Accept Only From*: If selected, users within this COS can only be monitored by the person or extension specified in the field associated with this radio button.

There are no special permissions for ShoreTel Contact Center agents or supervisors. They must have CoS with appropriate settings to enable contact center recording, monitoring, and barge in.

15.1.6 Reporting

As calls are received and handled by the agents of the contact center, records are kept to help supervisors manage the call flow and resources. Each call to the workgroup is logged in terms of how long it spends in the queue, how it ends, which agent handles the call, and how long the call takes.

Three different views of contact center activity are reportable at the ShoreTel server using these easy-to-run, standard reporting tools:

- Queue Summary Report
- Agent Summary Report
- Agent Detail Report

Please refer to Appendix C, starting on page 415 for more information on the workgroup reports.

15.2 Workgroup Navigation

To invoke the Workgroups list page, click the Workgroups link in the navigation frame, shown in Figure 15-2. From this list page, you can add a new workgroup or link to an existing workgroup.

ShoreTel™			Workgroups Help			
ShoreWare Director			Add new			
Logoff Administrator						
Administration						
• Users...						
• Trunks...						
• IP Phones...						
• Switches						
• Call Control...						
• Voice Mail...						
• Auto-Attendant Menus						
• Workgroups						
• Schedules						
• System Directory						
• Application Servers						
• Conference Bridges						
• Sites						
• System Parameters...						
• Preferences						
Name	Extension	Agents	Schedules			
			On-Hours	Holiday	Custom	
Sales	4806	4				
Special Promotions	4807	2				
Support Center	4803	5				
© 1998-2004 ShoreTel, Inc. All rights reserved.						

Figure 15-2 Workgroups List Page

The columns on the *Workgroups* list page are as follows:

- **Name:** This is the name of the workgroup.
- **Extension:** This is the workgroup extension.
- **# Agents:** This is the number of agents that are assigned to the workgroup. Maximum of 64 agents per workgroup or maximum of 16 when using Simultaneous Ringing.
Agents can be assigned to multiple workgroups.
- **On-Hours:** This is the name of the *On-Hours* schedule, if any, that is associated with a workgroup (named in the *Name* column at the left).

- **Holiday:** This is the name of the *Holiday* schedule, if any, that is associated with an existing workgroup.
- **Custom:** This is the name of the *Custom* schedule, if any, that is associated with an existing workgroup.

15.3 Adding and Editing Workgroups

This section explains how to add and edit workgroups from the *Workgroups* edit page. This page also lets you assign extensions, user group features, and configure schedules.

Workgroups
Edit Workgroup

[New](#) [Copy](#) [Save](#) [Delete](#) [Reset](#) [Help](#)

Edit this record [Refresh this page](#)

Name:

Extension:

Backup Extension: [Search](#)

DID: (DID Range: +14083313916 - 3939)

DNIS: [Edit DNIS Map](#)

User Group:

Mailbox (server) [Escalation Profiles and Other Mailbox Options](#)

Language:

Accept Broadcast Messages

Include in System Dial By Name Directory

Make Number Private

Recorded Name: [Record](#) [Play](#) [Erase](#) [Import](#) no audio input

Voice Mail Password: Confirm:

Enable Automatic Agent Logout on Ring No Answer

Workgroup Membership: [Edit Agents](#)

Workgroup Queue Handling: [Edit Queue Handling](#)

Wrap Up Time: Seconds

Current Call Handling Mode:

On-Hours
 Off-Hours
 Holiday
 Custom

Escalation Profile: [Edit this profile](#)

Schedule: [Edit this schedule](#)

Call Handling:

Distribution Pattern: Top Down Round Robin Longest Idle Simultaneous

Call Forward: Always No Answer/Busy

Always: Extension: [Search](#)
 External: (e.g. 9+1 (408) 331-3300)

Busy: Extension: [Search](#)
 External: (e.g. 9+1 (408) 331-3300)
 Queue

No Answer: Extension: [Search](#)
 External: (e.g. 9+1 (408) 331-3300)
 Queue

Logged Out: Extension: [Search](#)
 External: (e.g. 9+1 (408) 331-3300)
 Queue

Rings per Agent:

No Answer Number of Rings:

Mailbox:

Workgroup Greeting: [Record](#) [Play](#) [Erase](#) [Import](#) no audio input

Workgroup Assistant: [Search](#)

Enable Message Notification

Figure 15-3 Workgroups Edit Page

15.3.1 Parameters

The parameters on the *Workgroups* edit page are as follows:

- **Name:** This is the name of the workgroup.
This is a required parameter.
- **Extension:** This is the extension number of the workgroup.
If you change the extension number and there is an associated mailbox, messages will be retained.
This is a required parameter and must be unique. A maximum of 128 workgroup extensions can be defined.
- **Backup Extension:** This is the backup extension of the workgroup. If the workgroup does not answer after the specified number of rings (for example, server unavailable or network problem), the call is routed to this extension. This allows you to configure backup call routing in case of failures.
If the call volume is low, you can route these calls to an agent. If the call volume is high, it is recommended that you route these calls to a “must answer line” with a distinctive-sounding ring. Agents can use the call pickup feature to “pick” calls from the “must answer line” if the workgroup server is unavailable.
This is a required parameter.
- **DID:** This lets you assign a Direct Inward Dial (DID) number to the workgroup. You can assign one DID number to a workgroup.
This is an optional parameter.
- **DNIS:** The *Edit DNIS Map* button invokes the *Select DNIS Trunk Group* dialog box. This lets you select a trunk group for DNIS routing. Only trunk groups that are configured for DNIS will be presented in the dialog box. You can assign multiple DNIS numbers to a workgroup. Also refer to Section 15.4 on page 309.
DNIS is typically used to route 800-number calls to a workgroup or application.
This is an optional parameter.
- **User Group:** This lets you assign the workgroup to a user group from the accompanying drop-down list. The workgroup must inherit permission from the User Group’s CoS, since it has access to some telephony features (such as Trunk-to-Trunk Transfer and Call Forward External), as well as some voice mail features (such as Incoming Message Length and Message Notification). See Section 10.3 on page 185 and Section 10.2.1 on page 175.
If you are using **Barge In**, **Record**, and/or **Monitor**, make sure you have created user groups with the appropriate CoS permissions.
This is a required parameter.
- **Mailbox (server):** This provides the workgroup with a mailbox on the associated server. If you change the server, all messages are automatically moved to the new server.
Calls to the workgroup that go to voice mail end up in the workgroup mailbox. Calls to a user that go to voice mail end up in the user’s mailbox. The workgroup mailbox is shared by all workgroup members who run the Workgroup Agent or Supervisor Call Manager. In addition, you can log into the workgroup mailbox over the telephone using the mailbox number and voice mail password.

This is an optional parameter.

- **Language:** Select a language from the drop-down list. This is the language that will be used by the auto attendant for prompts played to calls in the queue, such as “your estimated wait time is...”
- **Accept Broadcast Messages:** This lets the workgroup receive broadcast messages. In general, you will want to remove the workgroup from the broadcast message list.

This is an optional parameter.

- **Include in System Dial By Name Directory:** This includes the workgroup in the auto-attendant’s dial-by-name directory.

This is an optional parameter.

- **Make Number Private:** Checking this check box removes this number from the system directory and call handling destination lists.
- **Recorded Name:** The **Record**, **Play**, **Enter**, and **Import** buttons let you record a name for the workgroup. This name is used as part of the default mailbox greeting as well as in the Dial By Name directory.

You can use your PC microphone and speakers or a telephone to play and record within ShoreWare Director. Please refer to the Auto-Attendant options for more information.

You can also import prompts into ShoreWare Director. Prompts must be recorded as μ -law, WAV files.

This is an optional parameter.

- **Voice Mail Password:** This lets you enter and confirm the workgroup’s voice mail password.

The voice mail password is used in conjunction with the mailbox number to log in to the workgroup mailbox over the telephone.

This is a required parameter. The initial default is “1234”. Passwords may be numeric only.

- **Enable Automatic Agent Logout On Ring No Answer:** This check box lets you automatically log out agents that do not answer a workgroup call after a specified number of rings.

This feature is useful to avoid calls repeatedly being offered to an agent who has physically left the workgroup but has forgotten to log out.

This is an optional parameter.

- **Workgroup Membership:** The *Edit Agents* button invokes the *Workgroup Membership* page, which lets you add and remove agents to and from the workgroup. You can also change the call distribution hunt order of agents via this page. See Figure 15-5.

This is a required parameter.

- **Workgroup Queue Handling:** The *Edit Queue Handling* button invokes the *Workgroup Queue Handling* edit page. This page lets you edit the *Queue Step Menu* as well as the *Queue Thresholds*. For more information, see Section 15.6 on page 310.

This is a required parameter.

- **Wrap Up Time:** **Wrap Up Time** allows agents a fixed period of time in seconds to complete post-call tasks before being presented with another call. When the call wrap-up time is set to zero, this feature is effectively disabled.

This is an optional parameter.

- **Current Call Handling Mode:** There is a read-only display of the current call handling mode as defined by any associated schedules.
- **Escalation Profile:** Select an Escalation Profile from the drop-down menu to associate the desired profile with this workgroup. (See Section 10.10.3 on page 239 for details.)
- **Schedule:** You can configure schedules against the **On-Hours**, **Holiday**, and **Custom** modes that automatically change the call handling of the workgroup. The rules for schedules are:
 - If it is custom time, use **Custom** mode;
 - If it is holiday time, use **Holiday** mode;
 - If it is on-hours time, use **On-Hours** mode;
 - Otherwise, use **Off-Hours** mode.

If *no schedules* are specified, *On-Hours* mode is used.

The **Edit this schedule** link provides a quick way to navigate to the associated schedule. For tips on editing schedules, refer to Chapter 14, starting on page 293.

This is an optional parameter.

- **Call Handling Distribution Pattern:** This lets you configure how calls are distributed to agents within the workgroup:
 - *Top Down* begins at the top of a list of agents and proceeds down the list looking for an available agent until one is found.
 - *Round Robin* selects the next agent on the list following the agent that received the last call and then starting again at the top of the list and so on. This works in a looping fashion.
 - *Longest Idle* distributes calls to the agent who has been idle the longest.
 - *Simultaneous* distributes calls simultaneously to all available agents; all phones ring at once.

This is a required parameter.

- **Call Forward:** These buttons let you specify when calls are forwarded. The conditions are *Always* or *No Answer/Busy*. The default is *No Answer/Busy*.

The *Always* condition forwards calls to the number specified in the *Always Destination* parameter immediately when a call is received.

The *No Answer/Busy* condition forwards calls:

- to the *No Answer Destination* after the specified number of rings, or
- to the *Busy Destination* immediately if the user's call stack is full.

This is a required parameter.

- **Always:** When the *Always* call forward option is selected, calls are forwarded immediately to this *Extension*. You can also forward calls to an *External* number (access code required). If the Call Forward parameter is set to *Always*, enter extensions here.

- **Busy:** When the *Busy* call forward option is selected, calls are forwarded to this extension immediately if all agents are busy. You can also forward calls to *Queue*, an *External* number (access code required), or *Extensions* here.
- **No Answer:** When the *No Answer* call forward option is selected, calls are forwarded to this extension after the specified number of rings. You can also forward calls to *Queue*, an *External* number (access code required), or *Extensions* here.
- **Logged Out:** When the *Logged Out* call forward option is selected, calls are forwarded to this extension if all agents are logged out of the workgroup. You can also forward calls to *Queue* or to an *External* number (access code required).
- **Rings per Agent:** This is the number of rings attempted before the call is forwarded to the next available agent.
- **No Answer Number of Rings:** This is the maximum number of rings that a call will ring before it is forwarded to the no-answer destination.
- **Mailbox / Workgroup Greeting:** The **Record**, **Play**, **Enter**, and **Import** buttons let you record a mailbox greeting for each call handling mode of the workgroup.
This is an optional parameter.
- **Workgroup Assistant:** To assign a workgroup assistant, select one from the drop-down list. When a caller is connected to the workgroup's voice mail and enters "0", the call is transferred to the workgroup assistant extension.
This is an optional parameter.
- **Enable Calling Message Notification:** This check box activates the message notification feature for the workgroup's mailbox.
This is an optional parameter.

15.4 Selecting a DNIS Trunk Group for DNIS Routing

To set DNIS routing, click *Edit DNIS Map* found on the *Workgroups* edit page. The DNIS Trunk Group dialog box (Figure 15-4) lets you select a trunk group for DNIS routing.

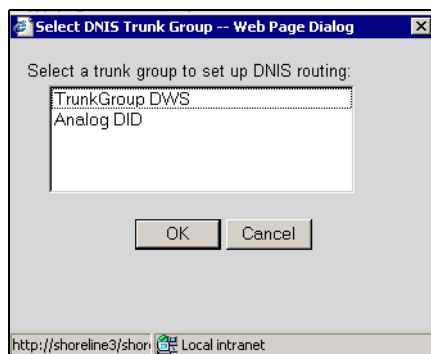


Figure 15-4 Select DNIS Trunk Group Dialog Box

Click an available trunk group from the list, and click OK.

15.5 Editing Workgroup Membership

To build a list of workgroup members, click *Edit Agents* found on the *Workgroups* edit page. The *Workgroup Membership* edit page (Figure 15-5) lets you add and remove agents to and from the workgroup.

Figure 15-5 Workgroup Membership Edit Page

This page lets you add and remove agents to and from the workgroup, and also change the order of agents. Note that if more members exist than can be displayed on one page, you can use the forward and back buttons to scroll through the members or enter filter criteria in the *Filter Users By* box. Also note that you can sort members by Extension, First Name, or Last Name by using the *Sort By* drop-down list.

To add members to a workgroup, select a member's name from list on the left side of the page and click *Add*. The member's name appears in the list on the right. When you add a new workgroup member, that member is in a logged out state by default.

To remove a member from the workgroup, select that member's name from the list at the right side of the page and click *Remove*. The member's name is returned to the list on the left side of the page.

To change the order of a member's hunt order in the workgroup, select the member's name in the **right-most** list and click *Move Up* or *Move Down* until the member appears in the order you want.

To change the member's login status, double-click the name from the **right-most** list and the *Edit Workgroup Member* dialog box (Figure 15-6) appears.

You can change the agent's *Client Type*, *Logged In*, or *Call Wrap Up* status, and click OK to commit the changes. The change is reflected in the *Name* list on the *Workgroup Membership* edit page.

15.6 Editing Workgroup Queue Handling

The *Workgroup Queue Handling* edit page (Figure 15-7) lets you edit the queue threshold and the Queue Step Menu. Click *Edit Queue Handling* from the *Workgroups* edit page.

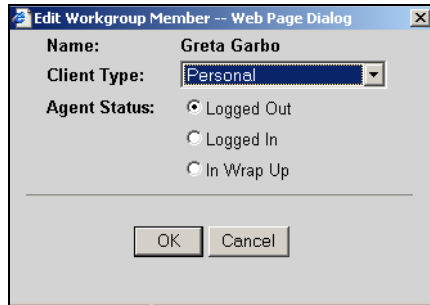


Figure 15-6 Edit Workgroup Member Dialog Box

A calling party that reaches the queue will hear the following, in order:

1. The recorded prompt.
2. Silence for eight seconds during the “wait for digits” time-out.
3. The estimated wait time (if enabled).
4. Music on hold for the configured duration.

The calling party does not have access to the DTMF actions while listening to music on hold.

The ShoreTel 6.1 system ignores DTMF actions if there is no message for the queue step.

The *Allow agents to pickup from queue* checkbox enables and disables the queue handling step menu.

A five-step menus can be set for each queue.

To set the Queue Step Menu:

Step 1 To set a step, click *Step 1*, *Step 2*, *Step 3*, *Step 4*, or *Last Step*. To skip a step, select *Skip This Step*.

Step 2 Select **Announce Estimated Wait Time** to enable this feature. This informs the caller of how much time is left in queue.

Estimated wait time is a moving average based on the duration of the previous calls and rounded to the minute. It is calculated as:

$$\text{Average wait seconds} = (("\text{Average wait seconds}" * 9) + "\text{New wait time}") / 10$$

Where the "New wait time" is the time it took the last call to get to an agent.

The wait time that is spoken to the caller is:

$$\text{Spoken wait time} = "\text{Position in queue}" * "\text{Average wait seconds}"$$

This means the wait time is based on a rolling weighted average of previous calls. After 10 calls, 61% of the time is based on the 10 most recent calls. After 20 calls, 86% of the time is based on the last 20 calls. The wait time may be inaccurate with a low call volume.

Step 3 Enter the value for the *Time Until Next Step* routine in the *Seconds* field.

Step 4 Enter the text used for the prompt in the **Prompt Text** field.

Step 5 Click **Record** to record the prompt. If you have a “canned” prompt, click **Import** to import the Wave file.

Figure 15-7 Workgroup Queue Handling Edit Page

Click *Play* to hear the recording. Click *Erase* to erase the recording.

Prompts on the ShoreTel 6.1 system can be imported into the system using μ -law, Wave file format. If you would like your prompts to match the voice of the ShoreTel 6.1 system, please contact Worldly Voices at www.worldlyvoices.com and request that “Connie” record your prompts. Worldly Voices provides this service with a rapid turnaround time for a nominal fee.

Step 6 Click an operation function from the **Operation** drop-down list and assign it an extension. The *Single Digit Actions* allow you to configure all the digits of the telephone keypad (0–9, #, *) for operations similar to an auto-attendant menu. The following actions are supported:

- None
- Repeat prompt
- Go to menu
- Transfer to extension
- Take a message
- Hang up

Details:

- For information about overflowing / interflowing calls to a workgroup call, see Section 15.6.1 on page 313.

15.6.1 Configuring Workgroup Overflow / Interflow

Workgroup overflow and interflow capabilities offer a way to reduce the wait time for callers who are dialing into an Automatic Call Distribution (ACD). This helps to ensure faster service and greater customer satisfaction.

Calls can be overflowed (i.e. transferred from one workgroup queue to another higher priority queue once a wait time threshold has been exceeded) in order to ensure that certain customers who have paid for a higher level of support service (e.g. "Gold Level") have their calls answered faster.

Alternatively, calls can be interflowed to any dialable number when a wait time threshold has been exceeded. This external number could be, for example, a cell phone number for a supervisor who could answer the call immediately.

Interflowing a call is typically done as the final step in a series of overflows. In other words, if a call is sent from one workgroup queue to another without being answered, the call is interflowed from the **Last Step** to an external number (such as the supervisor's cell phone).

To configure the Workgroup Overflow / Interflow feature:

- Step 1** The overflow / interflow functionality is set as the Last Step within the **Queue Step Menu**. Thus, you must follow the procedure in Section 15.6 on page 310 before configuring the overflow / interflow option.
- Step 2** Click on the **Last Step** tab and scroll down to the **Overflow / Interflow** section toward the bottom of the window, as shown below:
- Step 3** In the **Overflow / Interflow** section, select the desired radio button:
 - Selecting **None** results in the Overflow / Interflow behavior being disabled.
 - Selecting **Extension** enables the Overflow or Interflow behavior. If this is selected, click the Search button to locate the desired extension where calls will be overflowed or interflowed.
 - Selecting **External** enables the Interflow behavior. Enter the desired dialable number where calls are sent when the threshold is exceeded.
- Step 4** Select the **Maintain Wait Time** check box to store information about the length of time a caller has been waiting. The starting time associated with a call will be preserved even after the call has been transferred from one workgroup queue to another, and the caller's "place in line" will be reserved within the next queue. If the check box is disabled, the wait time will be lost when the caller is transferred and the caller will go to the back of the line in the new queue.

If the call is interflowed to an external number, the wait time cannot be retained. The system will recognize if these conditions are present and the **Maintain Wait Time** check box will be grayed out.
- Step 5** Click **Save** to store your changes.

Details:

- The number of seconds in the **Time Until Next Step** field for **Steps 1** through the **Last Step**, when added up, represent the total amount of time the caller waits before being overflowed / interflowed.

Workgroup Queue Handling [Save](#) [Reset](#) [Help](#)
 Edit Workgroup Queue Handling [Refresh this page](#) [Busy](#)

Queue Thresholds: [Edit Alert Thresholds](#)

Allow agents to pickup from queue

Queue Step Menu:

▶ Step 1 ▶ Step 2 ▶ Step 3 ▶ Step 4 ▼ Last Step

Skip This Step

Announce Estimated Wait Time

Time Until Next Step: Seconds

Prompt Text:

```
This is the last step of the HQ workgroup.
You will be overflowed to the RM workgroup in 15
seconds.
```

no audio input

Operation	Ext	Operation	Ext
0	---	6	---
1	888-558888	7	---
2	3333-555555	8	---
3	888-55309	9	---
4	---	*	---
5	---	#	---

Overflow / Interflow:

None:

Extension: [Search](#)

External: (e.g. 9+1 (408) 331-3300)

Maintain Wait Time

Figure 15-8 Overflow / Interflow options found at Last Step of WG Queue Handling

15.7 Workgroup Thresholds

To adjust the queue threshold, click *Edit Alert Thresholds* found on the Workgroup Queue Handling page. The *Workgroup Thresholds* page (Figure 15-9) appears.

Workgroup Thresholds [Save](#) [Reset](#) [Help](#)
 Edit Workgroup Queue Alert Thresholds [Refresh this page](#) [* modified](#)

Calls in Queue Warning: Calls

Calls Waiting Time Warning: Seconds

[Return to this workgroup's queue handling](#)

Figure 15-9 Workgroup Thresholds Page

The threshold of **Calls in Queue Warning** specifies the number of calls in queue that triggers an alert in the Queue Monitor in the workgroup Call Manager application.

The **Calls Waiting Time Warning** specifies the number of seconds that triggers an alert in the Queue Monitor for the longest wait time of any caller in the workgroup queue.

Managing the System Directory

This chapter discusses how the system directory is used and provides instructions for adding new entries and editing existing ones.

16.1 System Directory

The system directory is a company-wide address book that lists a user or outside contact by name and provides additional information, such as home address and phone numbers. This directory is read-only by general users through the ShoreWare clients; it can be updated by system administrators through ShoreWare Director. Users can copy system directory entries to their personal directories.

Individual users are added to the system directory when you add them to the system from the *Edit User* page (refer to Section 10.4 on page 187). You can also add new and edit existing entries from the *System Directory* page shown in Figure 16-1. In this way you can add commonly accessed outside contacts to the directory as well.

First Name	Last Name	Ext	Type	Site	Trunk Group	DID	Work	Home	Fax	Ce
Adam	Ant	4509	User Extension	Headquarters	Analog DID Group	+12133330001				
Betty	Boop	4510	User Extension	Headquarters						
Carl	Castle	4511	User Extension	Headquarters						
CC-IVR1		4821	Route Point	Headquarters						
CC-IVR2		4822	Route Point	Headquarters						
CC-IVR3		4823	Route Point	Headquarters						
CC-IVR4		4824	Route Point	Headquarters						
CC-IVR5		4825	Route Point	Headquarters						
Donald	Duck	4512	User Extension	Headquarters	Analog DID Group	+12133330002				
Ernie	Els	4513	User Extension	Headquarters	Analog DID Group	+12133330004				
Fred	Flintstone	4514	User Extension	Headquarters						
Greta	Garbo	4515	User Extension	Headquarters	Analog DID Group	+12133330005				
Howard	Hughes	4516	User Extension	Headquarters						
Ina	Island	4912	User Extension	Headquarters						
Johnathan	Seagull	4805	User Extension	Headquarters	Analog DID Group	+12133330003				
Kris	Kringle	4913	User Extension	Headquarters						
Les	Lesman	4914	User Extension	Headquarters						
Sales		4806	Workgroup	Headquarters						
Sales-JRN		4820	Route Point	Headquarters						
SMDI	ShoreDoc-HQ	4902	System User DN	Headquarters						
SMDI	ShoreDoc-HQ	4901	System User DN	Headquarters						
Special Promotions		4807	Workgroup	Headquarters						
Support Center		4803	Workgroup	Headquarters						
Support-JRN		4900	Route Point	Headquarters						
Tim	Toolman	4904	User Extension	Headquarters						

Figure 16-1 System Directory Page

ShoreWare Call Manager automatically populates each user's Quick Dialer with entries from the system directory, the user's personal directory, and all Microsoft Outlook Contact folders. This includes each user's personal contacts as well as any contacts on the Microsoft Exchange Server.

To access the system directory, click *System Directory* from the navigation frame.

The system directory can be filtered by site or dial number. You can also page through the directory by page, with a configurable number of records per page displayed.

The columns that appear in the table on the *System Directory* page are as follows:

- **First Name:** This is the first name of an existing directory entry.
- **Last Name:** This is the last name of an existing directory entry.
- **Ext.:** This is typically a contact's telephone number or extension.
- **Type:** This is the type of extension, such as workgroup, fax, and so on.
- **Site:** This is the site where the extension is located.
- **Trunk Group:** This is the trunk group associated with the extension.
- **DID:** This is the direct inward dial number for the user.
- **Work:** This is the work number for the user.
- **Home:** This is the home number for the user.
- **Fax:** This is the fax number for the user.
- **Cell:** This is the cell number for the user.

16.1.1 System Directory Entry Edit Page

To add new or edit existing user entries in the system directory, click the *Add new* link or an existing user first name in the table. The *System Directory Entry* edit entry page, shown in Figure 16-2, appears.

System Directory	
Edit User 4509	
New Copy Save Delete Reset Help	
Edit this record	Refresh this page
First Name:	<input type="text" value="Adam"/>
Last Name:	<input type="text" value="Ant"/>
Home Phone:	<input type="text"/>
Work Phone:	<input type="text"/>
Fax Phone:	<input type="text"/>
Cell Phone:	<input type="text"/>
Pager:	<input type="text"/>
E-mail Address:	<input type="text" value="AAnt@yourcompany.com"/>
Go to this user	

Figure 16-2 System Directory Edit Entry Page

16.1.2 Data-Entry Fields

Descriptions for each data-entry field on the *System Directory Edit Entry* page follow:

- **First Name:** This field displays the user's first name.
- **Last Name:** This field displays the user's last name.

If you are editing an existing user's information, these fields were automatically entered when you added the user on the Edit User page. If you modify this field on the System Directory edit entry page, it is reflected on the Edit User page and in all other lists.

Refer to Section 10.4 on page 187 for details.

- **Home Phone:** This is the user's home telephone number.
- **Work Phone:** This is a work telephone number for the user other than his or her extension. Do not enter a user extension number in this field.
- **Fax Phone:** This is the user's fax number.
- **Cell Phone:** This is the user's cellular telephone number.
- **Pager:** This is the user's pager number.
- **E-mail Address:** This is the user's e-mail address.

If you are editing an existing user's information, this field was automatically entered when you added the user on the Edit User page. If you modify this field on the System Directory edit entry page, it is reflected on the Edit User page and in all other lists.

For more information, see Section 10.4 on page 187.

16.1.2.1 Go to This User Link

The *Go to this user* link appears on the *System Directory* edit entry page when you are editing an existing user's directory entry. This link takes you to the *Edit User* page so that you can edit the user's general, personal, and distribution list options.

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 6.1 system.

ShoreTel 6.1 supports SIP trunks and SIP Extensions

17.1 SIP Trunks

Session Initiation Protocol (SIP - RFC 3261) is a newer protocol that is still being fine tuned by the IETF and that is regarded as having the potential to become the global signaling standard that will enable all switches, gateways, and phones to talk to one another.

The protocol, which works at the application layer, allows users to initiate interactive sessions between any network devices that support the protocol. SIP is capable of initiating or terminating Internet telephony calls and other multimedia applications such as video or gaming.

The protocol is based on a client-server model. With support for redirection services, networked users can initiate a call or receive a call, regardless of their physical location.

In its networking negotiations SIP takes into account the following pieces of information:

- the address of the end system
- the physical media
- the call recipient's acceptance to the invitation

The protocol then configures the parameters for the session and handles the call setup and tear-down.

SIP allows two discrete ShoreTel 6.1 systems to be integrated with any IP connection, without the need for physical tie trunking. (Note that care should be taken to make sure that the extension numbering plans in the two systems do not overlap, and that if they do overlap, translation tables need to be used to resolve conflicts.)

Further, the addition of SIP obviates the need to support other trunking standards, such as BRI, through use of a SIP gateway.

SIP trunks will be assigned to a particular switch as with any other trunk, so that SIP calls into and out of the ShoreTel 6.1 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.

17.1.1 Supported RFCs

ShoreTel 6.1 supports SIP Trunks as specified by the following RFCs:

- 1889 - Transport Protocol for RTP Applications
- 2806 - URL for Telephone Calls
- 2327 – Session Description Protocol (SDP)
- 2396 - URI (Uniform Resource Identifiers)
- 2833 - DTMF
- 2976 - SIP Info
- 3261 - SIP (Session Initiation Protocol)
- 3361 - DHCP (for IPv4)
- 3515 - SIP Refer Method
- 3891 - SIP Replaces Header
- 3892 - SIP Referred-by Mechanism
- 3966 - URI for Telephone Calls

17.1.2 General SIP Comments

Previous ShoreWare versions support SIP trunks in the following configurations:

- Static Trunking connects two PBXs within a ShoreWare installation
- Dynamic Trunking connects a PBX to a SIP device.
- Outbound SIP Trunking connects a PBX to an Ingate SIP Trunk Module, which can connect to external IP networks or to an ITSP to for connecting to the PSTN.

SIP extensions replaces the Dynamic SIP Trunking configuration available in releases previous to ShoreTel 8.1. ShoreTel 6.1 continues to support dynamic trunks configured in previous releases.

The SIP trunking implementation in previous versions does not support the use of SIP devices as ShoreWare extensions.

17.1.2.1 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.

17.1.2.2 DTMF

- ShoreTel 6.1 can be configured to use SIP INFO for DTMF signaling in environments where out-of-band DTMF is needed but RFC 2833 is not applicable. SIP tie trunks must use SIP INFO and cannot use RFC 2833 DTMF Relay.

- ShoreTel 6.1 supports RFC2833 (DTMF) for users calling over SIP trunks regardless of the negotiated voice codec.

17.1.2.3 Foreign Language Support

- In addition to English, ShoreTel 6.1 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.

17.1.2.4 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

17.1.2.5 Office Anywhere over SIP Trunks

ShoreWare supports Office Anywhere phone assignment to devices available over SIP trunks. Office Anywhere procedures and capabilities over SIP trunks is identical to that for reassignments to phones over other supported trunks.

Using Office Anywhere over SIP Trunks requires using an ITSP that supports SIP-INFO. If the ITSP does not support SIP-INFO, ShoreWare cannot support DTMF for users configured for the SIP Trunk who during Office Anywhere calls. This requires that they are configured for “Accept on Answer” and restricts them from using keypad features during the call. These users can still control their calls through Call Manager.

The following limitations apply:

- SIP Tie lines to other ShoreTel systems do not support Office Anywhere.
- SIP Trunks to supported SIP carriers through Ingate support Office Anywhere.
- SIP Trunks using SIP gateways supported by ShoreTel (including SIP-BRI 8 and Vegastream SIP gateways) support Office Anywhere.
- Gateways must support, and be configured to send, DTMF in INFO messages to support certain functions through Office Anywhere, including Wait for DTMF, ** flash, and ## hangup).
- Office Anywhere users cannot hear in-band call-waiting tone for a second call over SIP Trunks. Call Manager can signal the presence of a second call.
- Features not supported by normal calls on a SIP trunk are also not supported by Office Anywhere extension using a SIP trunk.

17.1.2.6 General Feature Limitations

- ShoreWare supports Music On Hold (MOH) over SIP trunks. The capacity limits of MOH switches is the same as other trunks; a switch can provide up to 15 streams. However, these streams can be to other switches or to SIP devices.
- If the ShoreTel server has a conference bridge 4.2 installed, you should not enable SIP. The conference bridge is not compatible with a ShoreTel 6.1 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.
- Silence detection on trunk-to-trunk transfers is not supported since it requires a physical trunk.
- Fax (and modem) redirection is not supported with SIP trunks as only physical trunks can detect fax tones.
- SIP devices cannot have a voicemail box, cannot use PCM, cannot be a member of a hunt group, cannot be a member of a workgroup, and will not appear in the dial by name directory of the AA because they are SIP trunks and not SIP extensions.

17.1.2.7 Additional Configuration Considerations

- SIP Info configuration in a Trunk Group should be enabled if ShoreTel SIP tie trunks are used.
- Overlapping number plans are not allowed between two systems tied with SIP trunks unless digit translation is used.
- When translating digits between two ShoreTel 6.1 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 6.1 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 6.1 is working with Dynamic Trunks:

- Multiple registrations of different numbers using the same IP address is not supported as ShoreTel 6.1 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 6.1 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 6.1 system.
- Director does not show information about the SIP devices registered in a switch. This information can be accessed by telnetting 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.
 - This parameter is found in Director under *Call Control > Options*

17.1.3 Configuration SIP Trunks

Configuring SIP on your ShoreTel 6.1 system consists of the following tasks:

- Configuring the ShoreTel 6.1 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.

17.1.3.1 Configuring the ShoreTel 6.1 System via Director

One of the first tasks involved in configuring your ShoreTel 6.1 system for Session Initiation Protocol (SIP) is to reserve the trunk. This can be done by following the procedure below:

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.

Switches
Edit ShoreGear-40/8 Switch [New](#) [Copy](#) [Save](#) [Delete](#) [Reset](#) [Help](#)

* modified

Edit this record [Refresh this page](#)

Name:

Description:

Site: Headquarters

IP Address: [Find Switches](#)

Ethernet Address:

Server to Manage Switch:

Caller's Emergency Service Identification (CESID): (e.g. +1 (408) 331-3300)

Music On Hold Source

Ports for the new switch will be created when you click 'Save'.

Port	<input type="checkbox"/> IP Phones	<input type="checkbox"/> Conference	<input checked="" type="checkbox"/> SIP Trunks	Description	Jack Number	Location
1	<input type="checkbox"/>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>
2	<input type="checkbox"/>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>
3	<input type="checkbox"/>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>
4	<input type="checkbox"/>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>
5	<input type="checkbox"/>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>
6	<input type="checkbox"/>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>
7	<input type="checkbox"/>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>
8	<input type="checkbox"/>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>
9	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>
10	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>

Figure 17-1 Configuring Switch for SIP

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.

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.

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 17-2 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 with a ShoreTel 6.1 system running a release previous to ShoreWare 8.1. 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 must have matching information in the associated fields, and the user ID and password of the device will be authenticated against the information stored in the ShoreTel 6.1 system.

Based on the authentication configuration:

- Disabled when set to “None”.
- Incoming calls are challenged when set to “Inbound-only”.
- When outbound SIP calls are challenged, credentials provided when set to “Outbound-Only”.
- All calls are authenticated when set to “all”.

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 6.1 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:

The screenshot shows a configuration window for a SIP trunk group. It is divided into several sections:

- Outbound:** A checked checkbox.
- Network Call Routing:**
 - Access Code: Input field containing '9'.
 - Local Area Code: Input field containing '408'.
 - Additional Local Area Codes: Edit button.
 - Nearby Area Codes: Edit button.
- Trunk Services:**
 - Local: Checked checkbox.
 - Long Distance: Checked checkbox.
 - International: Checked checkbox.
 - n11 (e.g. 411, 611, except 911 which is specified below): Checked checkbox.
 - 911: Checked checkbox.
 - Easy Recognizable Codes (ERC) (e.g. 800, 888, 900): Checked checkbox.
 - Explicit Carrier Selection (e.g. 1010xxx): Checked checkbox.
 - Operator Assisted (e.g. 0+): Checked checkbox.
 - Caller ID not blocked by default: Checked checkbox.
- Trunk Digit Manipulation:**
 - Remove leading 1 from 1+10D: Unchecked checkbox. Hint: Required for some long distance service providers.
 - Remove leading 1 for Local Area Codes (for all prefixes unless a specific local prefix list is provided below): Unchecked checkbox. Hint: Required for some local service providers with overlay area codes.
 - Dial 7 digits for Local Area Code (for all prefixes unless a specific local prefix list is provided below): Checked checkbox. Hint: Local prefixes required for some local service providers with mixed 7D and 1+10D in the same home area.
- Local Prefixes: Dropdown menu set to 'None' with a link 'Go to Local Prefixes List'.
- Prepend Dial Out Prefix: Empty input field.
- Off System Extensions: Edit button.
- Translation Table: Dropdown menu set to '<None>'.

Figure 17-3 Creating a SIP trunk group (outbound configurations)

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.)

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.

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.

The screenshot shows a web-based configuration interface for SIP trunks. At the top, there are buttons for 'New', 'Copy', 'Save', 'Delete', and 'Reset', along with a 'Help' link. Below these is a 'Refresh this page' button and a '* modified' indicator. The main form area is titled 'Edit this record' and contains the following fields:

- Site: Headquarters
- Trunk Group: SIP Trunk Group
- Name:
- Switch:
- SIP Trunk Type:
 - Dynamic
 - Use IP Address
- Number of Trunks (1 - 120):

Figure 17-4 Creating a SIP Trunk

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.
- **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.

17.1.3.2 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)

17.2 SIP Extensions

ShoreTel 6.1 supports the use of SIP devices that comply with RFC 3261 (Session Initiation Protocol) as ShoreWare extensions. SIP devices configured as ShoreWare extensions support most features offered by ShorePhones.

The ShoreTel 6.1 implementation of SIP devices support standards specified by the following documents:

- RFC 3261 Session Initiation protocol (SIP)
- RFC 2833 In-band DTMF/Out-of-band DTMF
- RFC 3515 SIP Refer Method
- RFC 3550 A Transport protocol for real-time (RTP/RTCP) applications
- RFC 3551 RTP Profile for Audio and Video Conferences with Minimal Control
- RFC 2327 Session Description Protocol (SDP)
- RFC 2396 Uniform Resource Identifiers (URIs)
- RFC 2806 URLs for Telephone Calls
- RFC 3966 URIs for Telephone Calls
- RFC 2976 SIP Info Method
- RFC 3264 SDP Offer/answer
- RFC 3842 Message waiting indication reception
- RFC 3265 Subscription for MWI events
- RFC 3891 The Session Initiation Protocol (SIP) “Replaces” Header
- RFC 4208 Session Timers in the Session Initiation Protocol

SIP extensions replaces the Dynamic SIP Trunking configuration available in previous ShoreWare releases. ShoreWare 8.1 continues to support dynamic trunks configured in previous releases.

17.2.1 Network Elements

Implementing SIP extensions require the following components:

- A **SIP Proxy** or **endpoint** is the logical entity to which SIP transactions communicate through a SIP device.

ShoreGear resources can be reallocated from supporting trunks, analog extensions, or IP devices to support SIP endpoints. Switch resources are reallocated to support the following number of endpoints:

- One trunk resource is reallocated to support 100 SIP endpoints.
- One analog extension resource is reallocated to support 100 SIP endpoints.
- One IP phone resource is reallocated to support 20 SIP endpoints.

ShoreGear half-width switches also provide built-in SIP proxy resources that do not require the reallocation of other resources.

- **Proxy servers** are network devices that facilitate communication between SIP endpoints. The Proxy Server forwards requests from endpoints to the correct endpoint or another server.

Within a ShoreTel 6.1 network, Proxy Server functionality is built into ShoreGear half width and full width switches. ShoreGear switches provide Proxy Servers for SIP devices within the same site as the switch. Administrators can designate up to

two Proxy Switches per site: one switch is assigned as the primary Proxy Server while the other switch acts as the backup Proxy Server.

One Virtual IP address (VIP) is designated as the SIP Proxy Server address for each site that supports SIP extensions. This address must be static and is assigned to the ShoreGear switch that is configured as the SIP proxy for the site. The VIP is moved to the backup Proxy Server switch if the primary Proxy Switch fails.

If only one Proxy Server is configured, a VIP address does not need to be configured.

- **Registrars** handle SIP REGISTER requests. Users send REGISTER requests for uploading their current location, which is then used by the Proxy server when routing information to the user.

The switch that acts as the Proxy Server also serves as the Registrar for the site.

- **SIP Call Managers** routes SIP calls to and from SIP endpoints. All switches acting as SIP Proxy Servers can also perform Call Manager functions.

ShoreWare automatically designates a SIP endpoints Call Manager when it registers. SIP endpoints send REGISTER requests to the ShoreGear switch serving as its proxy server. When an unprovisioned device registers, the switch forwards the request to ShoreWare Server IPCS, where it is authenticated. IPCS then assigns a Call Manager switch by load balancing across the site's switch resources. If the site does not have sufficient Call Manager resources, the SIP device is not provisioned.

When a call is made to or from an extension on the SIP device, the requests are sent to the Call Manager switch, which determines how the call is routed.

17.2.2 Supporting SIP Devices

17.2.2.1 Provisioning SIP Phones

The procedures for provisioning SIP extensions is similar to that for MGCP extensions:

1. The phone sends a REGISTER request to the Registrar server.

Existing registrations: The registrar server relays the request to SIP call manager switch, which then completes authentication.

New registrations: The request is relayed to IPCS, which then provisions the phone on a ShoreGear switch call manager.

2. The user inputs their ShoreWare username, SIP password, and SIP proxy address (VIP address) in the configuration form provided by the phone.

ShoreWare can recognize the extension, DID number, or the Director Login name as the user portion of a SIP Address of Record. The DID number is the preferred choice for creating the address of record.

3. If the device provides a URN, the system uses it to identify the device. If the device does not provide a URN, the system identifies the device by using the contact header information.

Changing the IP address of a SIP device may result in the listing of the device by two switches. This dual listing persists until the earlier registration expires. Therefore, identifying a device by the contact header may result in a dual switch listing if the contact header changes when the phone is provisioned.

All requests to the phone are routed through the SIP call manager switch directly to the phone or to the destination specified by the active call handling mode.

All requests from the phone are sent to the proxy server, which forwards them to the SIP call manager switch.

17.2.2.2 Phone Profiles

Various SIP device models support differing sets of features, including call control capabilities, codec compatibility, and provisioning procedures. SIP Profiles distinguish between the different SIP Device models, as described in the Section 17.2.6.3 on page 338.

17.2.2.3 Phone Management

The SIP Protocol specifies phone management levels to denote the degree to which the system maintains control over the configuration of the device. **Unmanaged devices** do not have their configuration managed by the phone system. **Managed devices** have their configuration tightly managed by the system.

ShoreTel 6.1 only supports unmanaged SIP devices.

17.2.2.4 Office Anywhere

Office Anywhere is a ShoreWare feature that assigns a ShoreWare extension to a ShorePhone or a calling device external to the system. Users transfer their extension to the phones upon which they perform the Office Anywhere transfer. The device from which the extension is transferred is designated as an **Anonymous Telephone** if it is a ShorePhone device.

When a user transfers an extension from a SIP device, ShoreWare recognizes the SIP device as an anonymous phone until the most recent REGISTER message sent by the phone expires. When the REGISTER message expires, the SIP device is removed from the IP phone list. While the SIP device is an Anonymous Telephone, the credentials of the user previously assigned to the device remain active.

An extension can be assigned to a SIP device through Office Anywhere only after the device becomes an Anonymous Telephone from an earlier Office Anywhere transfer and before the device is removed from the IP phone list. The only user that can use Office Anywhere to assign an extension to the device is the user that previously transferred the extension from the device.

17.2.3 User Features

This section describes the user features supported by SIP extensions.

17.2.3.1 Call Dialing and Initiation

SIP extensions support the following call initiation features:

- **Make Call:** Calls can be made from the phone or from Call Manager.
- **On hook dialing:** On hook dialing is supported from Call Manager.
- **Intercom:** Intercom calls are initiated from SIP devices by dialing *15 followed by #.

Intercom calls to SIP extensions are presented as regular calls.

- **Redial and Speed dial:** Redial and Speed dial initiated from SIP extensions through Call Manager operate similar to on hook dialing.

Redial and Speed dial methods differ for each SIP device model. Feature keys on specific SIP devices may be programmed to support speed dial.

- **E911:** Calls to emergency number 911 from SIP extension sends the user's emergency identification or CESID number with the call.

Unregistered SIP phones cannot dial 911 calls.

- **Dial plans and extension lengths:** SIP extensions support all extension lengths ShoreWare can configure. When the SIP call manager receives an Incomplete number or an illegally formed number from a SIP device, it returns the SIP response *484 Address Incomplete* or *420 Bad Address*, respectively.
- **Night bell:** SIP extensions can ring the night bell by dialing star code *14.

17.2.3.2 Call Handling

Call handling operations provide options for answering or routing incoming calls. SIP extensions support the following call handling options:

- **Answer call:** SIP extensions can answer calls only from the phone.

Offering calls can be redirected to Voice Mail, an Automated Attendant, or another extension through Call Manager.

- **Hang-up:** SIP extensions can hang up calls from the phone or from Call Manager.
- **Ring No Answer (RNA):** The number of rings that trigger a No Answer response is specified in the Call Handling Mode definition for each user. When the No Answer condition is triggered, the SIP call manager redirects the call to the RNA destination as specified by Director.
- **Busy:** If the user call stack size is smaller than the phone call stack size, calls that overflow the user call stack are redirected to the busy destination as specified by Director.

When the user call stack is larger than the phone call stack, the SIP phone reject overflow calls with SIP response *486 Busy*. The switch then redirects the call to the busy destination as specified by Director.

- **Forward Always:** SIP extensions supports Forward Always. When this parameter is set, all calls will be forwarded to the destination specified by Director.
- **Call waiting:** The specific call waiting implementation differs for each SIP phone model. SIP extensions support call waiting to one or multiple simultaneously offered calls for SIP devices that support this feature.
- **Call rejection:** When the SIP phone rejects the call with *603 Decline* response code, the switch fails the call and play the reorder tone to the phone
- **Call redirect:** If the SIP phone returns *3xx* response code, the switch redirects the call to the user's RNA destination. If RNA destination is not configured, the reorder tone is played.
- **Find Me:** SIP extensions support FindMe and Call Message notification.

17.2.3.3 Caller ID

Caller ID is the caller information transmitted to the other party during a voice call.

- **Caller ID presentation:** SIP extensions can display caller name and number.

- **Caller ID blocking:** SIP extensions support CID blocking and Make Number Private.
- **Caller ID for Workgroup and Hunt Group agents:** The system sends the name of the Hunt Group or Workgroup with the original caller number while the call rings. After the call is answered, the system sends the original caller name and number.

17.2.3.4 Call Control

Users perform call control operations to manage their active voice calls. SIP extensions support the following call control operations:

- **Hold:** Call hold and unhold are performed from the phone. Implementation of the reminder ring for held calls differs among SIP phone models. This feature is handled internally by the phone.
- **Basic transfer:** SIP extensions support blind transfers that use REFER. Transfers using re-INVITE are not supported.
- **Consultative transfer:** SIP extensions support consultative transfers that use REFER. Transfers using re-INVITE are not supported.
- **Park from SIP phone:** Calls are parked from SIP extensions by dialing *11 followed by #.
- **Unpark on SIP phone:** SIP phone users pickup calls parked from their extension stack by dialing *12 followed by their extension number.
Taking the SIP phone offhook is not sufficient to resume a call parked on the same extension.
- **Pickup:** SIP extensions pickup calls by dialing *13 followed by #.
- **Unpark:** SIP extensions unpark calls by dialing *12 followed by #.
- **Conference Calls:** Three party conference calls initiated from the phone use the phone's resident MCU.
 - Three party conferences calls initiated from Call Manager uses MakeMe conferencing.
 - MakeMe conferencing is used when a SIP phone joins a conference call.
 - Four to six party conference calls are supported using MakeMe and must be initiated through Call Manager.
- **Call recording:** SIP Extension call recordings are supported only for only calls involving a physical trunk and can be initiated only through Call Manager.
- **Voicemail:** SIP extensions reach voice mail by dialing # on the phone or pressing the Call Manager VM button.
- **MWI:** SIP extensions support Message Warning Indicator by using NOTIFY or SUBSCRIBE/NOTIFY on phone models that support MWI. MWI is configurable through SIP phone profiles.
- **Agents:** SIP extensions are available for Workgroup and Hunt Group agents.
- **Bridged call appearance:** SIP extensions do not support Bridged Call Appearances.
- **DTMF:** SIP Extensions support DTMF tones as specified by RFC 2833.
- **Huntgroup busyout:** SIP extensions can busyout huntgroups by dialing *18 followed by HG#.

17.2.3.5 Specialty Calls

Specialty calls establish communication between system users during voice calls.

- **Silent monitor:** SIP Extensions cannot initiate this operation. Silent Monitor cannot be performed on SIP extension phones.
- **Barge In:** SIP Extensions cannot initiate this operation. Barge In cannot be performed on SIP extension phones.
- **Whisper Page:** SIP Extensions cannot initiate this operation. Silent Monitor cannot be performed on SIP extension phones.

17.2.4 System Features

SIP extensions support the following system features:

- **Account codes:** Users on SIP extension can be forced to use account codes for external calls.
- **Bandwidth allocation:** ShoreWare allocates bandwidth for SIP voice calls.
- **Backup Auto Attendant (BAA):** The SIP call manager switch provides BAA to the SIP extension. All BAA prompts are played in G.711 format.
- **Supported Codecs:** ShoreWare default settings support negotiation of the following codecs:
 - L16/16000
 - L16/8000
 - AAC_LC/32000
 - PCMU/8000
 - PCMA/8000
 - G722/8000
 - DVI4/8000
 - BV32/16000
 - BV16/8000
 - G729/8000

New codecs can be added to the Supported Codec List. This list is accessed from Director by selecting *Administration* -> *Call Control* -> *Supported Codecs* from the main menu.

- **Fax redirection:** Fax calls to SIP extensions are redirected to the site fax redirect number. SIP extensions do not support T.38.
- **Music on Hold (MoH):** ShoreWare does not provide music on hold to SIP extensions
- **Office anywhere – external devices:** SIP extension user can be configured such that Office Anywhere can be performed from devices external to the ShoreWare system.
- **On-Net dialing:** SIP extensions can use on-net dialing.
- **PSTN failover:** SIP extensions can be reached via PSTN failover.
- **Packetization period:** The default packetization period for all calls involving SIP extensions is 20 ms.
- **Video call:** The *Allow intersite video calls* setting restricts video calls across sites. ShoreWare does not allocate bandwidth for video calls.

- **Country Call Progress Tones and Ring Tones:** SIP extensions provide call progress and ring tones for countries supported by ShoreWare.
- **Language support:** SIP extensions provide support for languages as required by the countries supported by ShoreWare.

17.2.5 User Assignment

ShoreWare can use the extension, DID, or Client User ID to identify or authenticate a user.

The contact name, or the information that is placed in the SIP header for messages from the device, is constructed from the contact name configured in the SIP device.

The address of record is the IP address by which a user can be reached through the SIP device. The address of record is from the user name configured in the SIP device.

17.2.6 Configuration

This section describes the procedures required to implement SIP extensions.

17.2.6.1 Specifying the SIP Network Elements

The Edit Sites panel supports the following SIP Network configuration tasks for the specified site:

- Defines the IP address for the SIP Proxy and Registrar Server.
- Designates the ShoreGear switches that serve as the site's SIP Proxy and Registrar servers. Data entry fields are available for designating primary servers and secondary servers

To configure the SIP Network Elements for a ShoreWare site:

Step 1 Open the Sites list by selecting *Administration -> Sites* in the Director menu.

Step 2 Open the Edit Site panel, as shown in Figure 17-5, for the desired site by clicking that site's link located in the Site column.

The screenshot shows the 'Edit Site' panel for a site named 'Headquarters'. The panel is divided into several sections. At the top, there are buttons for 'New', 'Copy', 'Save', 'Delete', and 'Reset', along with a 'Help' link. Below these is a 'Refresh this page' button. The main configuration area includes fields for 'Name' (Headquarters), 'Country' (United States of America), and 'Language' (English(US)). A red arrow points to the 'SIP Proxy' section, which contains fields for 'Virtual IP Address' (10.23.15.106), 'Proxy Switch 1' (HQ-120-New), and 'Proxy Switch 2' (None). Below this is an 'Emergency Number List' section with a checkbox for 'Trunk Access Code Required' and a field containing '911'. At the bottom, there is a link for 'Edit IP Phone Address Map' and a copyright notice: '© 1998-2007 ShoreTel, Inc. All rights reserved.'

Figure 17-5 Configuring SIP Network Elements on the Edit Site panel

The SIP Proxy section, located in the bottom section of the panel, configures the following parameter settings:

- **Virtual IP Address:** This parameter defines the IP address of the site's SIP Proxy Server and Registrar server. The IP address is independent of the switch that performs the server functions. SIP extensions require that this parameter is set to a valid address.
- **Proxy Switch 1:** This setting designates the switch that performs the site's SIP server functions. The drop down menu lists all switches assigned to the site. SIP extensions require the setting of this parameter.
- **Proxy Switch 2:** This setting designates the switch that performs the site's SIP server functions when the switch specified by Proxy Switch 1 is not available. This parameter is optional.

17.2.6.2 Setting the SIP Call Controls

The Edit Call Control Options panel configure SIP parameters for the ShoreWare system.

To configure the SIP Network Elements for a ShoreWare system:

- Step 1** Open the Edit Call Control Options panel, as shown in Figure 17-6, by selecting *Administration* -> *Call Control* -> *Options* from the Director menu.

Figure 17-6 Editing SIP Parameters in the Call Control Options panel

Step 2 Set values for the following call control parameters:

- **Realm:** A realm is a protection domain over which SIP authentication parameters are specified. For digest authentication, each such domain defines a set of usernames and passwords, through which access is granted. The Realm parameter configures the name of the realm to which the site devices belong.

Resetting the Realm requires a system reboot.

- **Enable SIP Session Timer:** SIP Session Timer measures the timeout period that determines when devices transmit or receive a RE-INVITE or UPDATE

method to refresh a session still in progress. This timeout informs stateful proxies that a call remains active.

When *Enable SIP Session Timer* is selected, the following parameters are active:

- *Session Interval*: This parameter specifies the keep alive period for SIP Sessions when *Enable SIP Session Timer* is enabled.
- *Refresher*: This parameter designates the device – the UAC or UAS – that sends the refresh. The method (RE-INVITE or UPDATE) is dynamically selected based on the Timer header supported by the SIP endpoint.

17.2.6.3 SIP Profiles

A SIP Profile is a ShoreWare record, defined for a specific SIP device, that lists characteristics, properties, features, and settings for that device. ShoreWare uses SIP Profiles to monitor and service the SIP devices connected to the system. ShoreWare supports **predefined** and **user defined** profiles.

- **Predefined profiles** are provided by ShoreWare to support generic devices or devices for which a specific profile is not defined. Although predefined profiles cannot be deleted or modified, they can be deactivated or superseded by user defined profiles.
- **User defined profiles** are created through Director and list parameter settings for specific SIP device models.

SIP Profile List

The SIP Profiles List panel, shown in Figure 17-7, displays the names of all SIP Profiles on the system. To access the SIP Profile list panel, select *Administration -> Call Control -> SIP Profiles* from the Director menu.

Each row corresponds to a SIP Profile. Table columns list the following profile parameters:

- **Name**: This parameter is the label by which Director refers to the profile.
- **User Agent**: This parameter is the expression ShoreWare uses to identify devices covered by the profile. ShoreWare compares this expression to the User Agent field in the header of SIP packets handled by the system.
- **Enabled**: This parameter lists the status of the profile. ShoreWare uses only profiles that are enabled when evaluating the characteristic set of a device.
- **Priority**: This parameter determines the order by which the profiles are evaluated against the SIP packet header. Profiles with larger priority values are evaluated before profiles with smaller priority values.

The User Agent field of successive profiles are compared to the User-Agent field of the SIP packet header until a match is found. The profile containing the matching User-Agent field is then used to specify device configuration settings.

To add a profile, press the New button in the top left corner of the panel.

To remove a profile, select the checkbox that corresponds to the profile to be deleted, then press the Delete button in the upper left corner of the panel. Predefined profiles cannot be deleted.

To edit a profile, open the Edit SIP Profile panel by clicking the name of the profile to be modified.

<input type="checkbox"/>	Name	User Agent	Enabled	Priority
<input type="checkbox"/>	ShorePhone IP8000	ShoreTel/ST_PH1_2.6.0 (3)	Yes	50
<input type="checkbox"/>	System	*	Yes	10
<input type="checkbox"/>	Cisco-7960	Cisco-CP7960G/7.5	Yes	100

© 1998-2007 ShoreTel, Inc. All rights reserved.

Figure 17-7 SIP Profiles List panel

Edit SIP Profile

The Edit SIP Profiles panel, shown in Figure 17-8, is the Director list panel from which parameter settings for the specified profile are configured. The Edit SIP Profile panel is accessed from the SIP Profile List by adding a new profile or editing an existing profile. All parameter fields can be edited for User Defined profiles. *Enable* is the only parameter that can be modified on predefined panels.

Sip Profile Edit Sip Profile [New](#) [Copy](#) [Save](#) [Delete](#) [Reset](#) [Help](#)

Edit this record [Refresh this page](#)

Name:

User Agent:

Priority:

Enable

System Parameters:
 OptionsPing=1
 SendEarlyMedia=0
 MWI=none
 ICodecAnswer=0
 StripVideoCodec=0

Custom Parameters:

© 1998-2007 ShoreTel, Inc. All rights reserved.

Figure 17-8 Edit SIP Profile panel

Parameters listed on the panel include:

- **Name:** This parameter is the label by which Director refers to the profile. The field contents are not editable for predefined profiles.
- **User Agent:** This parameter is the index key that ShoreWare compares to the User-Agent field in the header of inbound REGISTER or INVITE methods handled by the system. The field contents are not editable for predefined profiles.
- **Priority:** This parameter is used to determine the profile that is used when the User Agent field of multiple profiles match the User-Agent field in the SIP packet header. The field contents are not editable for predefined profiles.

- **Enable:** This parameter lists the status of the profile. ShoreWare uses only profiles that are enabled when evaluating the characteristic set of a device.
- **System Parameters:** This field lists the device characteristics and default settings.
- **Custom Parameters:** The contents of this field list additional device settings or overwrite default settings listed in the System Parameters field. Supported parameters include:
 - *optionsping*: Set this parameter to 1 if the SIP device can process SIP OPTIONS command. This command is sent to the SIP device as Keepalive message.
 - *sendearlymedia*: Set this parameter is 1 when the SIP device is capable of receiving “early media”. Some BAA prompts are streamed as early media.
 - *mwi*:
 - * Set to *none* if the SIP device does not support MWI
 - * Set to *subscribe* when the SIP device subscribes for message waiting service.
 - * Set to *notify* when the SIP device will can receive MWI notification without subscribing for the service.
 - *ICodecAnswer*: When set to 1, only one codec is set in answer SDP.
 - *StripVideoCodec*: When set to 1, ShoreSIP UA strips video codec from SIP SDP.
 - *AddGracePeriod*: When extra time needs to be added to the expire time for SIP registrations
 - *FakeDeclineAsRedirect*: When set to 1, response code *603 decline from SIP endpoint* is treated as redirect to CHM destination.

17.2.6.4 Allocating Switch Resources

A SIP extension can be utilized only if an unused SIP Proxy Port is available on a switch assigned to the site. Proxy ports are allocated through the Edit Switch panel. Figure 17-9 displays the Edit ShoreGear 50 Switch panel.

ShoreGear switches provide two SIP proxy port sources: Built-in capacity and Port assignment

- **Built-in capacity:** ShoreGear half-width switches, such as the SG-50, provide IP phone, SIP trunk, and SIP proxy resources that are independent of port switches. The number of resources varies with each switch model. Each resource unit supports one IP phone, one SIP trunk, or five SIP Proxies.

The panel in Figure 17-9 indicates that the SG-50 provides 20 resource units. The resources allocation configured in Figure 17-9 is:

- 10 resources as IP phones
- 5 resources as SIP trunks
- 5 resources that provide 100 SIP proxy ports.

To configure the Built-in resource for SIP Proxy Port allocation, enter the desired number of IP Phone and SIP Trunk resources in the specified data entry fields. The remaining resources from the total available are automatically designated for use as SIP proxy ports.

- **Port resources:** Switch ports can be configured to support 100 SIP proxy ports.

To configure a port for SIP Proxy Port allocation, access the drop down menu of the desired port in the port section of the panel and select the 100 SIP Phones option.

Switches
Edit ShoreGear 50 Switch

Buttons: New, Copy, Save, Delete, Reset

Refresh this page

Name: RM Kila 50

Description:

Site: Remote Site

IP Address: 10.23.87.117 Find Switches

Ethernet Address: 00-10-49-07-27-DC

Server to Manage Switch: RM DVM

Caller's Emergency Service Identification (CESID): (e.g. +1 (408) 331-3300)

Built-in Capacity: IP Phone + SIP Trunk = Total
10 + 5 = 15 of 20 (100 SIP proxy ports)

Music On Hold Source
Music On Hold Gain (-49 to 13): 0 dB

Use Analog Extension Ports as DID Trunks

RM Kila 50

Port	Port Type	Trunk Group	Description	Jack Number	Location
1	Available		P01		
2	100 SIP Proxy		P02		
3	5 IP Phones		P03		
4	Available		P04		
11	Available		P05		
12	Available		P06		

© 1998-2007 ShoreTel, Inc. All rights reserved.

Figure 17-9 Allocating SIP Proxy Ports

17.2.6.5 Setting an Extension Password

An extension is enabled for SIP if a value is assigned to the User's SIP Password parameter. The SIP Password parameter is located at the bottom of the Edit User: General panel, as shown in Figure 17-10. Clearing the SIP Password data fields disables the extension from supporting SIP.

17.2.6.6 Monitoring SIP Phones

All SIP extension devices on the ShoreWare network are listed on the IP Phone panel in Director, as shown in Figure 17-11. SIP devices are deleted from this panel when they are physically removed from the network.

Clicking on the name of a SIP device displays an Edit IP Phone panel for that device. The Edit IP Phone panel displays, in addition to the information listed on the IP Phones list panel, the Credential Name, User Name, Contact, and Address of Record for the device. The only information editable from this panel is the name of the device.

SIP Register popup

The SIP Register popup, shown in Figure 17-12, displays registration information for the corresponding device. This popup is accessed by pressing the SIP Registration on the IP Phones panel.

SIP devices can be manually registered using SIP Registration button on the IPPhones panel.

The screenshot shows the 'Users' administration interface. At the top, there are buttons for 'New', 'Copy', 'Save', 'Delete', and 'Reset', along with a 'Help' link. Below these are navigation tabs: 'General' (selected), 'Personal Options', 'Distribution Lists', and 'Workgroups'. A 'Refresh this page' link is also present. The form contains several input fields: 'First Name' (Jane), 'Last Name' (Smith), 'Number' (33708), 'Client User ID' (JSmith), 'Client Password' (two masked fields), 'Voice Mail Password' (two masked fields), 'SIP Password' (two masked fields, highlighted with a red arrow), 'Email Address' (JSmith@yourcompany.com), 'Conference Bridge' (Server dropdown set to 'None', User ID, Password), and 'IM Server' (Server dropdown set to 'None', IM ID). A footer note reads '© 1998-2007 ShoreTel, Inc. All rights reserved.'

Figure 17-10 Setting a SIP Password

The screenshot shows the 'IP Phones' administration page. It includes a 'Filter IP Phones' section with 'Find Now' and 'Clear' buttons, and dropdowns for 'By Sites' (All Sites) and 'By Switches' (All Switches). Below is the 'IP Phone List' with a search filter '1: 00-10-49-02-6E-6B - SIP-C540E51A8105F1' and pagination controls showing '14 Records' and '25 per page'. A table lists the devices with columns: Name, Site, Switch, MAC Address, IP Address, User (Current and Home), Phone Type, Assign To, and Expiration. A footer note reads '© 1998-2007 ShoreTel, Inc. All rights reserved.'

Name	Site	Switch	MAC Address	IP Address	User	Phone Type	Assign To	Expiration
00-10-49-02-6E-6B	Headquarters	HQ-120-New	00-10-49-02-6E-6B	10.23.17.103		IP110		
00-10-49-02-71-93	Headquarters	HQ-120-New	00-10-49-02-71-93	10.23.107.100		BB24 Button Box		
00-10-49-03-8B-B8	Remote Site	RM 40/80	00-10-49-03-8B-B8	10.23.17.105	Remote-GSmith	IP212		
00-10-49-03-DC-4F	Remote Site	RM 40/80	00-10-49-03-DC-4F	10.23.17.106	Remote-GSmith	IP230		
00-10-49-0C-B6-16	Remote Site	RM 40/80	00-10-49-0C-B6-16	10.23.17.104	Richard Smith	IP560		
33719	Headquarters	HQ2 - 60/12		10.23.15.107:5060	HssUA SIP	_System		
SIP-329400048E433C40914FAD26C3EBEE69	Remote Site	RM 40/80		10.23.16.100:5060	33722	Cisco-7960		
SIP-54153808C832E745BA8DF4498B38D163	Remote Site	RM 40/80		10.23.16.100:5060		Cisco-7960		8/21/2007 11:49:41 AM
SIP-653099ED0E70834EA95B74B6E679B05B	Headquarters	HQ-120-New		10.23.17.101:5062		_System		8/14/2007 4:24:04 PM
SIP-7463D91B6574A140BC3AB44953C0E17D	Headquarters	HQ2 - 60/12		10.23.17.101:5062	ksmith	_System		
SIP-A1D6784A2E575549BA168C483C69F4CF	Remote Site	RM 40/80		10.23.17.100:5060	DSmith	Cisco-7960		
SIP-C540E51A8105F140A0CCE36864275D72	Remote Site	RM 40/80		10.23.17.100:5060		Cisco-7960		8/21/2007 11:52:19 AM

Figure 17-11 SIP devices listed on IP Phones list panel

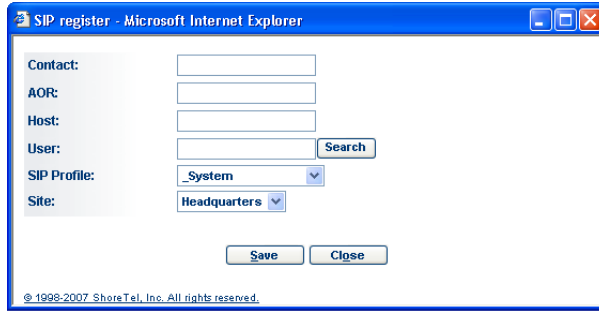


Figure 17-12 SIP Register popup panel

Maintenance

This chapter provides information about the maintenance of the ShoreTel 6.1 system. The sections in this chapter are:

- Overview
- Quick Look
- Switch Maintenance
- Switch Connectivity
- Event Log
- Services
- Event Filters

18.1 Overview

The ShoreTel 6.1 system provides maintenance information through ShoreWare Director, so you can check the operational status of a switch, port, server, or service. It also provides information to resolve an event or error that might occur while the system is running. Operational status and events are tracked by the system. The information is displayed on the maintenance pages that you access from the *Maintenance* link in the navigation frame, as shown in Figure 18-1.

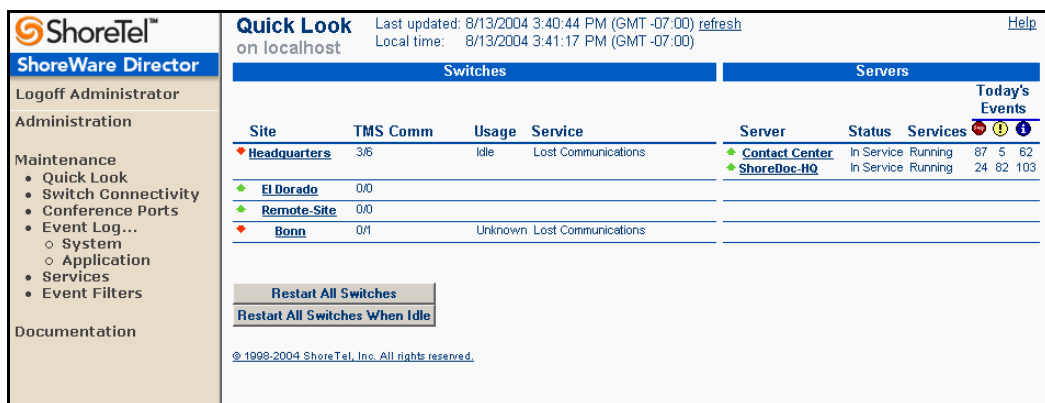


Figure 18-1 Maintenance Link in Navigation Frame

Maintenance pages include:

- Quick Look
- Switch Maintenance
- Switch Connectivity
- Event Log
- Services

- Event Filters

18.2 Quick Look

The *Quick Look* maintenance page, shown in Figure 18-2, provides a snapshot of the entire ShoreTel 6.1 system. It includes information about each site and the corresponding switches and servers.

Quick Look Last updated: 5/30/2008 10:51:41 AM (GMT -07:00) [Refresh](#) [Help](#)
 on 10.23.44.102 Local time: 5/30/2008 10:52:06 AM (GMT -07:00)

Switches				Servers			
Site	TMS Comm	Usage	Service	Server	Status	Services Used	Disk
♦ Headquarters	4/5	Idle	Lost Communications	♦ Headquarters	In Service	Running	17 % 2 9 33
♦ Remote	2/2	Idle	In Service	♦ RemoteSS	In Service	Running	5 % 4 1 73

Apply This Command to All Switches:

© 1998-2007 ShoreTel, Inc. All rights reserved.

Figure 18-2 Quick Look Page

The *Quick Look* page is described as follows:

- **Refresh:** Click this word, located at top of the panel, right of the *Last Updated* time, to update the panel information. The *Quick Look* panel automatically updates every 60 seconds.
- **Switches:** This area lists all of the sites on your system. Sites without servers are listed beneath their associated site that has a server. The following is represented under *Switches*:
 - **Site:** Clicking the name of a site will bring you to the *Maintenance Switches Summary* page. See Figure 18-3.
 - **TMS Comm:** This summarizes the communication state of all switches at the site. The first number represents switches with which TMS can currently communicate. The second number is the total number of switches at the site.
 - **Usage:** Shows either an Idle or In Use state.
 - **Service:** This summarizes the service state of all the switches at the site.
- **Servers:** This area lists all of the ShoreWare servers.
 - **Server:** Sites that have a server list the server in bold and a link to the *Quick Look Server Maintenance* page is provided. Sites without servers list the associated server they are served by.
 - **Status:** This summarizes the status of the server.
 - **Services:** This summarizes the status of the ShoreWare services on the server.
 - **Today's Events:** This summarizes the events that have been recorded in the event log on the server. See Section 18.3.9.3 on page 358.
- **Apply this Command to All Switches:** Each option initiates restart operations for all system switches. The restart begins immediately when an option is selected:
 - **Restart:** Select this option to immediately shut down, then restart, all switches. Calls that the switches are servicing when this command is selected are lost.

- **Restart when idle:** Select this option to shut down, then restart, each switch as as the calls they are currently servicing are completed. Calls that are in place when this command is selected are allow to completely normally.
- **Reboot:** This option performs the same operation as Restart.
- **Reboot when idle:** This option performs the same operation as Reboot when idle.

18.2.1 Maintenance Switches Summary

The *Maintenance Switches Summary* page (shown in Figure 18-3) is accessed by clicking on a site name listed on the *Quick Look* page. The switches summary lists all ShoreGear voice switches configured at the site. Switches are identified by name and IP address.

Maintenance - Switches Summary Last updated: 7/13/2007 5:45:22 PM (GMT -07:00) [refresh](#)
Local time: 7/13/2007 5:45:28 PM (GMT -07:00) [Help](#)

on 10.23.26.100 for site HeadQuarters [Go to Quick Look page](#)

Switch	IP Phones	SIP Trunks	Bridged Call Appearances	Type	IP Address	MAC Address	Comm	Conference	Hunt Groups	Usage	Service	Command
SHARED ATI PRI Switch	0/0	0/0	0/24	T1	10.23.79.201	00-10-49-04-A2-CA	5/10	0/0	0	In-Use	In Service	Select...
T1 tie	0/0	0/0	0/24	T1	10.86.26.104	00-10-49-00-05-18	0/10	0/0	0	Unknown	Lost Communication	Select...
E-911 to Area sw	0/0	0/0	0/24	T1	10.86.26.103	00-10-48-00-11-77	0/10	0/0	0	Unknown	Lost Communication	Select...
XO PRI switch	0/0	0/0	0/24	T1	10.86.0.200	00-10-49-00-1E-B1	0/10	0/0	0	Unknown	Lost Communication	Select...

Figure 18-3 Maintenance Switches Summary Page

The **Restart All** and **Restart All When Idle** buttons restart all switches. When clicked, **Restart All** restarts all switches and discontinues any calls in progress. Clicking **Restart All When Idle** restarts each switch only after all calls have ended.

The columns on the *Switches Summary* page are as follows:

- **Switch:** Clicking a switch name invokes the **Switch Maintenance** page for that switch. Clicking a SoftSwitch name invokes the *Server Maintenance* page. See Figure 18-14.
- **IP Phones:** This column lists the number of IP phones served by the switch and the IP phone capacity of the switch, according to the number of ports set to support IP phones. This information is presented as a ratio: IP phones/capacity.
- **SIP Trunks:** This column lists the number of SIP trunks served by the switch and the SIP capacity of the switch, according to the number of ports set to support SIP trunks. This information is presented as a ratio: SIP trunks/capacity.
- **Bridged Call Appearances:** This column lists the number of BCA extensions served by the switch and the BCA capacity of the switch, according to the number of ports set up to support BCA extensions. This information is presented as a ratio: BCA extensions/capacity.
- **Type:** This displays the type of this voice switch:
 - *120/24:* ShoreGear-120/24 voice switch
 - *24:* ShoreGear-24 voice switch
 - *T1:* ShoreGear-T1 voice switch
 - *SW:* SoftSwitch on any server
 - *60/12:* ShoreGear-60/12 voice switch
 - *40/8:* ShoreGear-40/8 voice switch
 - *8:* ShoreGear-8 voice switch

- *E1*: ShoreGear-E1 voice switch (outside USA only)
- **IP Address**: This is the switch's IP address.
- **MAC Address**: This is the switch's Ethernet address.
- **Comm**: This indicates the number of voice switches with which this switch is currently communicating. This is expressed as a fraction (# switches communicating /# total switches).
Clicking the *Comm* fraction takes you to the *Switch Connectivity* page, which provides more detail on each switch's communication status. See Figure 18-17.
- **Conference**: Indicates the number of conference ports currently in use.
- **Hunt Groups**: Indicates the number of hunt groups on the switch.
- **Usage**: This is the usage state of the switch.
 - *Unknown*—The usage state is unknown. This might be true because communication between the server and the switch might have been lost.
 - *In Use*—At least one port or IP phone has an active call.
 - *Off Hook*—At least one port or IP phone is off-hook, but no ports are in use.
 - *Idle*—There are no ports or IP phones off-hook or in-use.
- **Service**: This is the service state of the switch.
 - *Lost Communication*—The server lost communication with the voice switch. Note that the voice switch may be fully operational but the ShoreWare server cannot see the voice switch due to a networking issue. This also occurs when the voice switch is powered off.
 - *Upgrade in Progress*—The voice switch is currently being upgraded with a new software version.
 - *Restart Pending*—A *Restart when idle* command was issued but the restart did not occur because ports are still in use.
 - *Firmware Version Mismatch*—The voice switch is running a version of software that does not match the version on the server. The voice switch continues to run call control but does not have access to any voice services on the server. This typically happens on software upgrades after the server was upgraded but before the voice switches were restarted and upgraded. Note that voice switches at the same firmware version also continue to operate together.
 - *Firmware Update Available*—This means the server has a new optional version of firmware available for the voice switch. A voice switch in this state continues to run call control as well as access the voice services on the server. This state typically happens when you install a patch on the ShoreWare server. If you want, or need, the patch to propagate to the voice switches, you must restart them.
 - *FTP Booted*—This means the ShoreGear voice switch did not boot from FLASH memory but booted from an FTP server, most likely on the ShoreWare server. You can correct this problem by rebooting the voice switch. If this does not correct the problem, please contact ShoreTel's Customer Response Center.
 - *Port Out of Service*—This means all ports on the ShoreGear voice switch are out of service.
 - *IP Phone(s) Out of Service*—This means one or more IP phones associated with the switch are out of service.

- **Port Out of Service**—This means one or more, but not all, ports or IP phones are out of service on the ShoreGear voice switch. Ports or IP phones typically go out of service because either someone manually put them out of service or the call control software automatically put them out of service due to a signaling problem (for example, the dial tone was not received from the central office).
- **In Service**—This means the configured ports or IP phones are ready for service.

The **Command** drop-down list displays a list of commands, as shown in Figure 18-4. These commands let you change the service status of voice switch.

Maintenance - Switches Summary Last updated: 11/10/2005 4:47:42 PM (GMT -08:00) [refresh](#)
 on 10.2.29.27 for site Headquarters Local time: 11/10/2005 4:48:23 PM (GMT -08:00) [Help](#)
[Go to Quick Look page](#)

Switch	IP Phones	SIP Trunks	Type	IP Address	MAC Address	Comm	Conference	Hunt Groups	Usage	Service	Command
ATT PRM switch	0/0	0/0	T1	10.86.10.13	00-10-49-BB-1E-D5	0/13	0/0	0	Unknown	Lost Communication	Select...
Att T1	0/0	0/0	T1	10.86.10.11	00-10-49-00-14-B9	0/13	0/0	0	Unknown	Lost Communication	Select... Restart Restart when idle Put in service Put out of service Put out of service when idle
OSE T1	0/0	0/0	T1	10.86.29.17	00-10-49-00-05-18	0/13	0/0	0	Unknown	Lost Communication	Put in service Put out of service Put out of service when idle

Figure 18-4 Switch Operation Commands

The service state of the switch can be changed with the following commands:

- **Restart:** This restarts the voice switch. Active calls are dropped.
This is a forceful way to shed traffic from a voice switch when you are performing a software upgrade.
 - **Restart when idle:** This restarts the voice switch once all ports are idle. All idle ports are put out of service and remaining ports are put out of service when they go idle. Once all ports are out of service, the voice switch restarts.
This is a graceful way to shed traffic from a voice switch when you are performing a software upgrade. Note that active calls are completed but no new calls can be made or received until the voice switch restarts.
 - **Put in service:** This puts all ports on the voice switch in service. Ports already in service with active calls are not affected.
 - **Put out of service:** This places all ports on the voice switch out of service. Active calls are dropped.
This is a forceful way to shed traffic from a voice switch when you need to replace the switch.
 - **Put out of service when idle:** All idle ports are put out of service and remaining ports are put out of service when they go idle.
This is a graceful way to shed traffic from a voice switch when you need to replace the switch.
- SoftSwitches only have *Restart* and *Restart When Idle* in the command list.

18.3 Switch Maintenance

Clicking an individual switch name on the *Switches Summary* page brings you to the associated *Switch Maintenance* page. The individual *Switch Maintenance* pages are very similar but may vary to some degree. Examples of Switch Maintenance sub-section are shown on the pages that follow. Each type of switch may show less than all of the details show.

Switch Maintenance

ShoreGear-120/24 Doc-24

[Go to Maintenance - Switches Summary page](#)

Last updated: 4/14/2004 1:43:57 PM (GMT -07:00) [refresh](#)

Local time: 4/14/2004 1:44:08 PM (GMT -07:00)

[Help](#)

Switch

Host name	IP Address	MAC Address	Usage	Service	Command
◆ Doc-24	10.7.0.30	00-10-49-00-0D-52	Idle	In Service	Select...

Category	Service
Fan	◆ OK
Temperature	◆ OK
Talk Battery Voltage	◆ OK
Ring Voltage	◆ OK

Ports

Port	Description	Usage	Service	Command
◆ 1	Johnathan Seagull	Idle	In Service	Select...
◆ 6	ADID-0	Idle	In Service	Select...
◆ 8	1234	Idle	In Service	Select...
◆ 10	Betty Boop	Idle	In Service	Select...
◆ 11	Carl Castle	Idle	In Service	Select...
◆ 13	Ernie Els	Idle	In Service	Select...
◆ 14	Fred Flintstone	Idle	In Service	Select...
◆ 15	Greta Garbo	Idle	In Service	Select...
◆ 16	Howard Hughes	Idle	In Service	Select...

IP Phones

IP Phones	Phones	Usage	Service
◆ IP Phones Maintenance	1/5	Idle	In Service

Details

(Date Format: mm/dd/yyyy)

Last Boot Time:	4/14/2004 9:47:11 AM (GMT -07:00)
Boot Source:	Flash
Connect Time:	4/14/2004 9:46:33 AM (GMT -07:00)
Boot ROM Version:	2.0.8.5
Firmware Version:	9.2.8000.0
CPU Board Version:	Fab 6 (8 DSP IPBX-24)
Telephony Board Version:	Rev C
Ethernet Negotiation:	100M Half Duplex (Link)

Figure 18-5 Switch Section of the Switch Maintenance Page

18.3.1 Switch Section

The top of the *Switch Maintenance* page repeats some of the information on the *Switches Summary* page. The *Switch Maintenance* page also expose the individual ports as well as more detailed information on the switch operation.

The following is additional state information for ShoreGear-120/24, ShoreGear-60/12, ShoreGear-40/8, ShoreGear-T1, and ShoreGear-E1:

- Fan: Status of the fan, with the following states:
 - OK

- Slow
- Failed
- Unknown

Switches: 24, 12, 8, T1, E1

- **Temperature:** Status of the temperature, with the following states:

- OK
- Yellow Alarm
- Red Alarm
- Unknown

Switches: 24, 12, 8, T1, E1

- **Talk Battery Voltage:** Status of the *Talk Battery Voltage*, with the following states:

- OK
- Failed
- Unknown

Switches: 24, 12, 8

- **Ring Voltage:** Status of the Ring Voltage, with the following states:

- OK
- Failed
- Unknown

Switches: 24, 12, 8

18.3.2 Link Section

The ShoreGear-T1 and ShoreGear-E1 uses T1, T1 PRI, or E1 PRI signaling and additional information is presented on the status of the link and on the status of the PRI D-Channel.

Figure 18-6 displays the Switch Maintenance panel for a ShoreGear T1 switch.

Figure 18-7 displays the Switch Maintenance panel for a ShoreGear 220E1 switch.

The following is the link state information for the SG-T1/E1 and SG-220T1/220E1:

- **D-Channel (PRI only)**
 - *States:* In Service, Out of Service, Unknown
 - *Commands:* Reset, Put in service, Put out of service, Put out of service when idle
- **Line Coding**
 - *States:* OK, Bipolar Violations, Loss of Signal, Unknown
- **Framing**
 - *States:* OK, Yellow Alarm, Bit Error, Out of Frame, Unknown
- **Loopback**
 - *States:* Off, On, Unknown
 - *Commands:* Turn On, Turn Off
- **Span**
 - *Commands:* Reset, Put in service, Put out of service, Put out of service when idle.

Switch Maintenance Last updated: 5/18/2004 4:06:30 PM (GMT -07:00) [refresh](#) [Help](#)
 ShoreGear-T1 First T1 Local time: 5/18/2004 4:06:52 PM (GMT -07:00)

[Go to Maintenance - Switches Summary page](#)

Switch

Host name	IP Address	MAC Address	Usage	Service	Command
First T1	10.7.0.244	00-10-49-00-07-6C	Unknown	Lost Communication	Select...

Category **Service**

Fan	Unknown
Temperature	Unknown

Link

Category	Status	Command
Line Coding	Unknown	
Framing	Unknown	
Loopback	Unknown	Select...

Performance **Past 15 Minutes** **Past 24 Hours**

Error Free Seconds	Unknown	Unknown
Errored Seconds	Unknown	Unknown
Severely Errored Seconds	Unknown	Unknown
Unavailable Seconds	Unknown	Unknown
Out of Frame	Unknown	Unknown

Channels

Channel	Description	Usage	Service	Command
---------	-------------	-------	---------	---------

Details (Date Format: mm/dd/yyyy)

Last Boot Time:	Unknown
Boot Source:	Unknown
Last Connect Time:	Unknown (GMT -07:00)
Boot ROM Version:	0.0.0.0
Firmware Version:	0.0.0.0
CPU Board Version:	
CPU Board FPGA Version:	
Ethernet Negotiation:	Unknown

Figure 18-6 **Link Section of the Switch Maintenance Page (SG-T1)**

The following is the link performance information for the SG-T1/E1 and SG-220T1/220E1:

- **Error Free Seconds:** The number of error-free seconds that occurred in the last 15 minutes and 24 hours.
- **Errored Seconds:** The number of errored seconds that occurred in the last 15 minutes and 24 hours.
- **Severely Errored Seconds:** The number of severely errored seconds that occurred in the last 15 minutes and 24 hours.
- **Unavailable Seconds:** The number of seconds the server was not available.
- **Bit Error Rate:** Not used.
- **Out of Frame:** The number of times the link has been out of frame in the past 15 minutes and 24 hours.

Switch Maintenance Last updated: 2007-6-21 13:54:06 (GMT -07:00) [Refresh](#) [Help](#)
 ShoreGear 220E1 220E1 switch Local time: 6/21/2007 1:54:44 PM (GMT -07:00)

[Go to Maintenance - Switch Summary](#)

Switch						
Host name	IP Address	MAC Address	Usage	Service	Command	
220E1 switch	10.86.0.4	00-10-49-00-00-04	Unknown	Lost Communication	Select...	

Category	Service
Fan	Unknown
Temperature	Unknown

Link						
Category	Status	Command	Performance		Past 15 Minutes	Past 24 Hours
D-Channel	Down		Error Free Seconds	0	0	
Line Coding	Unknown		Errored Seconds	0	0	
Framing	Unknown		Severely Errored Seconds	0	0	
Loopback	Off	Select...	Unavailable Seconds	0	0	
Span		Select...	Out of Frame	0	0	

Channels						
Channel	Usage	Service	Command			
			Reset			
			Put in service			
			Put out of service			
			Put out of service when idle			

Figure 18-7 Link Section of the Switch Maintenance Page (SG-220E1)

18.3.3 Ports Section

Each Switch Maintenance page has a *Ports* section, where the configured ports are shown. The *Ports* section shows the port number, port description (first and last name or trunk name), usage state, service state, and available commands. Refer to Figure 18-8.

The following are the port **Usage** states:

- **Unknown:** The state is unknown, likely because communication between the server and the switch has been lost.
- **In Use:** The port has at least one active call.
- **Off Hook:** The port is off-hook and has no active calls.

The following are the port **Service** states:

- **In Service (green arrow up):** The port is in service.
- **Out of Service (red arrow down):** The port is out of service.

On telephone ports, outbound calls cannot be made (no dial tone) and inbound calls will not be offered.

On trunk ports, outbound calls will not seize the trunk, and inbound calls are not answered. On loop start trunks, the trunk is seized to emulate a busy condition to the central office.

- **Out of Service (operational) (red arrow down):** The port is out of service due to a manual “put out of service” command.

Applies only to trunks. The trunk failed to seize. The switch automatically attempts to seize the trunk on a periodic basis. When successful, the trunk is automatically put back in service.

The following are the port **Commands**:

Switch Maintenance Last updated: 11/10/2005 1:03:18 PM (GMT -08:00) [refresh](#)
 ShoreGear-120/24 101 Fuji-24 Local time: 11/10/2005 1:03:27 PM (GMT -08:00) [Help](#)

[Go to Maintenance - Switch Summary](#)

Switch						
Host name	IP Address	MAC Address	Usage	Service	Command	
101 Fuji-24	10.2.29.20	00-10-49-00-22-A4	Off-Hook	IP Phone(s) Out of Service	Select...	

Category	Service
Fan	OK
Temperature	OK
Voltages	OK

Ports				
Port	Description	Usage	Service	Command
1	SMDI	Idle	In Service	Select...
3	SMDI	Idle	In Service	Select...
4	HQ AP100	Idle	In Service	Select...
5	HQ Fax	Idle	In Service	Select...
6	HQ Analog	Idle	In Service	Select...

Hunt Groups				
Extension	Description	Usage	Service	Command
818-5999	Newest Hunt Group 5999	Idle	Normal	Select...

IP Phones			
IP Phones	Phones	Usage	Service
IP Phones Maintenance	45/45	Idle	IP Phone(s) Out of Service (Operational)

SIP Trunks			
SIP Trunks	Trunks	Usage	Service
SIP Trunks Maintenance	1/10	Idle	In Service

Details	
(Date Format: mm/dd/yyyy)	
Last Boot Time:	11/9/2005 2:06:27 PM (GMT -08:00)
Boot Source:	Flash
Connect Time:	11/9/2005 2:05:45 PM (GMT -08:00)
Boot ROM Version:	8.2.230.0
Firmware Version:	11.6.2303.0
CPU Board Version:	Fab 2 (6 DSP SG-24)
Link Status:	100M Full Duplex (Auto)
Active Interface:	LAN1

Figure 18-8 Ports, Hunt Groups, and Details Sections of Switch Maintenance Page

- **Reset:** Resets the port (puts the port out of service then automatically back in service). Active calls are dropped.
- **Put in service:** Puts the port back in service.
- **Put out of service:** Puts the port out of service. Active calls are dropped.
- **Put out of service when idle:** Puts an idle port out of service and puts an in-use port out of service when it goes idle.

18.3.4 Hunt Groups Section

Double-clicking on a Hunt Group on the Switch Maintenance Page shows the *Hunt Groups* edit page. You can use the *Command* drop-down to busy out a hunt group or return it to normal service from this page

This section shows the number of Hunt Groups configured on the Switch, The information provided in the *Hunt Groups* section includes:

- **Extension:** The extension number of the hunt group.
- **Description:** The text name of the hunt group.

Hunt Groups				
Extension	Description	Usage	Service	Command
4909	DoctorsGroup	Idle	Unknown	Select...
Details (Date Format: mm/dd/yyyy) <ul style="list-style-type: none"> Select... Make Normal Make Busy 				

Figure 18-9 Changing State of a Hunt Group

- **Usage:** The current usage state of the hunt group: Idle or Normal
- **Service:** Current service state of the hunt group: In Service, Out of Service, Unknown
- **Command:** Choose between Normal or Make Busy. Selecting Make Busy idles out the Hunt Group.

18.3.5 IP Phone Maintenance

The ShoreGear-120/24 *Switch Maintenance* screen has an additional section for IP phones. It shows how many IP phones are connected through the switch and the capacity of the switch. ShoreGear-120/24 switches can support as many as 120 IP phones, the ShoreGear 60/12 can support up to 60 IP phones, and the ShoreGear-40/8 can support up to 40 IP phones.

- **Phones:** Lists the number of IP phones on the switch.
- **Usage:** The current usage state of the IP phones: Idle or Normal
- **Service:** Current service state of the IP phones: In Service, Out of Service, Unknown

You can perform several functions on the IP phones connected to the switch by double-clicking the *IP Phones Maintenance* link. Figure 18-11 shows the *IP Phones Maintenance* page.

You can view the service history of a phone by clicking on its name from the IP Phone Maintenance page.

18.3.6 SIP Trunks Section

The *Switch Maintenance* screen has an additional section for SIP trunks. This section shows how many SIP trunks are connected through the switch and the capacity of the switch.

- **Trunks:** Lists the number of SIP trunks on the switch.
- **Usage:** The current usage state of the trunk group: Idle or Normal
- **Service:** Current service state of the trunk group: In Service, Out of Service, Unknown

18.3.7 Details Section

The following additional information is provided in the *Details* section:

- **Last Boot Time:** The last time the switch booted.
- **Boot Source:** Flash, FTP boot, or unknown boot source.

Switch Maintenance Last updated: 4/14/2004 1:43:57 PM (GMT -07:00) [refresh](#) [Help](#)
 ShoreGear-120/24 Doc-24 Local time: 4/14/2004 1:44:08 PM (GMT -07:00)
[Go to Maintenance - Switches Summary page](#)

Switch

Host name	IP Address	MAC Address	Usage	Service	Command
Doc-24	10.7.0.30	00-10-49-00-0D-52	Idle	In Service	Select...

Category

Category	Service
Fan	OK
Temperature	OK
Talk Battery Voltage	OK
Ring Voltage	OK

Ports

Port	Description	Usage	Service	Command
1	Johnathan Seagull	Idle	In Service	Select...
6	ADID-0	Idle	In Service	Select...
8	1234	Idle	In Service	Select...
10	Betty Boop	Idle	In Service	Select...
11	Carl Castle	Idle	In Service	Select...
13	Ernie Els	Idle	In Service	Select...
14	Fred Flintstone	Idle	In Service	Select...
15	Greta Garbo	Idle	In Service	Select...
16	Howard Hughes	Idle	In Service	Select...

IP Phones

IP Phones	Phones	Usage	Service
IP Phones Maintenance	1/5	Idle	In Service

Details (Date Format: mm/dd/yyyy)

Last Boot Time: 4/14/2004 9:47:11 AM (GMT -07:00)
 Boot Source: Flash
 Connect Time: 4/14/2004 9:46:33 AM (GMT -07:00)

Boot ROM Version: 2.0.8.5
 Firmware Version: 9.2.8000.0
 CPU Board Version: Fab 6 (8 DSP IPBX-24)
 Telephony Board Version: Rev C

Ethernet Negotiation: 100M Half Duplex (Link)

Figure 18-10 IP Phones Section of Switch Maintenance Page

Switch Maintenance Last updated: 4/14/2004 2:42:55 PM (GMT -07:00) [refresh](#) [Help](#)
 IP Phones Local time: 4/14/2004 2:43:22 PM (GMT -07:00)
[Go to Switch Maintenance](#)

IP Phone List

Apply this command to all selected IP Phones:

<input type="checkbox"/>	Name	Site	MAC Address	IP Address	Usage	Service	Current	Home	Firmware	Boot ROM	Phone Type
<input type="checkbox"/>	ShoreDoc IP Phone	Headquarters	00-04-F2-00-8E-B9	10.7.0.40	Idle	In Service	Adam Ant	Adam Ant	MGCP v1.2.4	2.4.1	IP-100

Figure 18-11 IP Phones Maintenance Page

- **Connect Time:** The last time the server re-established a connection with the switch.
- **Boot ROM Version:** The boot ROM version number. Recommended 2.0.x.x or higher.

SIP Trunks			
SIP Trunks	Trunks	Usage	Service
SIP Trunks Maintenance	1/10	Idle	In Service

Figure 18-12 SIP Trunks Maintenance Page

- **Firmware Version:** The firmware version number the voice switch is running.
- **CPU Board Version:** The version number of the switch's CPU board.
- **Telephony Board Version:** The version number of the telephony board.
- **Ethernet Negotiation:** The rate at which the voice switch negotiated the Ethernet interface. Ethernet states include:
 - 10: Half-Duplex
 - 10: Full-Duplex
 - 100: Half
 - 100: Full

18.3.8 ShoreGear-T1 PRI and ShoreGear-E1 PRI Maintenance

When you click the name of a ShoreGear-T1 switch configured for PRI of the ShoreGear-E1 on the *Switches Summary* page, the *Switch Maintenance* page shown in Figure 18-13 appears.

18.3.9 Quick Look Server Maintenance—Main Server

Click a server from the list on the *Quick Look Server Maintenance* page. The main server is shown in Figure 18-14. It is divided into three sections: *Status*, *Services*, and *Events*.

18.3.9.1 Quick Look Server Status Area

The information displayed in the *Status* areas is as follows:

- TAPI States:
 - *OK*—The TSP on the main server knows that the applications on the server have logged into TAPI
 - *Lost TAPI*—The TSP on the main server knows that the applications on the server are not logged into TAPI.
- SMTP Send States:
 - *OK*—The SMTP messages for the server are leaving.
 - *Failed*—The SMTP outbound queue on the server is not sending e-mail messages. The SMTP transport is used to send voice messages between voice mail servers.

18.3.9.2 Quick Look Server Services Area

The *Services* area provides details about the ShoreWare services areas shown in Table 18-1.

Clicking the ShoreWare service name takes you to a page use for starting or stopping that service, as shown in Figure 18-15.

Switch Maintenance
 ShoreGear-T1
 ShorelineT1PRI-1

Last updated: Mon May 14 18:42:49 [refresh](#)
 Local time: Mon May 14 18:42:53

[Click here or press F1 for help](#)

Switch

Host name	IP Address	Usage	Service	Command
ShorelineT1PRI-1	10.3.0.21	Idle	In Service	Select...

Category **Service**

Fan Ok

Temperature OK

Link

Category	Status	Command
Line Coding	Ok	
Framing	Ok	
Line Loopback	Off	Select...
Payload Loopback	Off	Select...

Performance

	Past 15 Minutes	Past 24 Hours
Error Free Second	0	0
Errored Second	0	0
Severely Errored Second	0	0
Unavailable Second	0	0
Bit Error Rate	0	0
Out of Frame	0	0

Channels

Channel	Description	Usage	Service	Command
1	4085551212	Idle	In Service	Select...
2	4085551212 (2)	Idle	In Service	Select...
3	4085551212 (3)	Idle	In Service	Select...
4	4085551212 (4)	Idle	In Service	Select...
5	4085551212 (5)	Idle	In Service	Select...
6	4085551212 (6)	Idle	In Service	Select...
7	4085551212 (7)	Idle	In Service	Select...
8	4085551212 (8)	Idle	In Service	Select...
9	4085551212 (9)	Idle	In Service	Select...
10	4085551212 (10)	Idle	In Service	Select...

Details

Last Boot Time: 5/14/2001 8:42:53 PM

Boot Source: Flash

Connect Time: 5/14/2001 8:42:59 PM

Boot ROM Version: 2.0.8.5

Firmware Version: 2.3.42.0 (Service Pack 1)

CPU Board Version: Fab 2 (6 DSP IPBX-T1)

CPU Board FPGA Version: 14

Ethernet Negotiation: 100M Full Duplex (Auto)

Figure 18-13 ShoreGear-T1 PRI or ShoreGear-E1 PRI Switch Maintenance Page

18.3.9.3 Quick Look Server Events

The *Quick Look Events* area provides a quick summary of event activity and lists all events that have been logged during the last two days—*Today* and *Yesterday*. Clicking the *Application Log* icon takes you to the *Application Event Log* maintenance page. See Section 18.6 on page 362 for more information.

The information displayed in the *Quick Look Events* area includes:

- **Date:** The day the event was reported: Today or Yesterday.
- **Errors:** The number of errors that were reported. Errors require your immediate attention.
- **Warnings:** The number of warning messages that were reported. Warnings alert you to potential errors.

Main Server Maintenance Last updated: 4/14/2004 3:16:13 PM (GMT -07:00) [refresh](#)
 on localhost Local time: 4/14/2004 3:16:50 PM (GMT -07:00) [Help](#)

[Go to Maintenance - Switches Summary page](#)

Status

Name	Status
TAPI	OK
SMTP Send	OK

Services [Go to All Services](#)

Service	Description	Status	Command
ShoreTel-ACC	ShoreWare Account Code Collection Server	Running	Select...
ShoreTel-CDR	ShoreWare Call Accounting	Running	Select...
ShoreTel-CSISSVC	ShoreWare CSIS Server	Running	Select...
ShoreTel-DRS	ShoreWare Distributed Routing Service	Running	Select...
ShoreTel-DTASSvc	ShoreWare Distributed Telephony Application Service	Running	Select...
ShoreTel-EventWatch	ShoreWare Event Watch Server	Running	Select...
ShoreTel-IPCS	ShoreWare IP-Phone Configuration Service	Running	Select...
ShoreTel-IPDS	ShoreWare IP-Phone Display Service	Running	Select...
ShoreTel-MailServ	ShoreWare Voice Mail Message Server	Running	Select...
ShoreTel-PortMgr	ShoreWare Voice Mail Port Manager	Running	Select...
ShoreTel-SoftSwitch	ShoreWare Software Telephony Switch	Running	Select...
ShoreTel-TMS	ShoreWare Telephony Management Server	Running	Select...
ShoreTel-VMail	ShoreWare Voice Mail Application	Running	Select...
ShoreTel-WGSvc	ShoreWare Workgroup Server	Running	Select...
ShoreTel-Zin	ShoreWare Database Management Service	Running	Select...

Events [Go to Application Log](#) [Go to System Log](#)

Date	Errors	Warnings	Informational
Today	21	37	153
Yesterday	-	-	5

Figure 18-14 Quick Look Main Server Maintenance

Name	Description	Details
ShoreTel-ACC	ShoreWare Account Codes Collection Server	This service collects call information by account code and reports call activity in the call detail reports.
ShoreTel-CDR	ShoreWare Call Accounting	This service records call accounting information, call queuing data, and media stream data.
ShoreTel-CSISSVC	ShoreWare CSIS Server	This service manages communications between the server and the Call Manager.
ShoreTel-DTASSvc	ShoreWare Distributed Telephony Application Service	This service provides connectivity between applications and instances of distributed TMS.
ShoreTel-DRS	ShoreWare Distributed Routing Service	This service allows the ShoreTel system to scale beyond 60 switches.
ShoreTel-EventWatch	ShoreWare EventWatch Server	This service monitors the event log and delivers e-mail notifications on certain events.
ShoreTel-IPCS	ShoreWare IP Phone Configuration Service	This service automatically configures IP phones as they are inserted into the system.
ShoreTel-IPDS	ShoreWare IP Phone Display Service	This service manages the IP phone display.
ShoreTel-MailServ	ShoreWare Voice Mail Message Server	This service is part of the ShoreWare Voice Mail system.

Table 18-1 ShoreWare Service Areas

Name	Description	Details
ShoreTel-PortMgr	ShoreWare Voice Mail Port Manager	This service is part of the ShoreWare Voice Mail system.
ShoreTel-SoftSwitch	ShoreWare Software Telephony Switch	This is the ShoreWare call control software running the server generally used to host virtual users.
ShoreTel-TMS	ShoreWare Telephony Management Server	This service provides the telephony platform for ShoreWare applications and services.
ShoreTel-Vmail	ShoreWare Voice Mail Application	This service is part of the ShoreWare Voice Mail system.
ShoreTel-WGSvc	ShoreWare Workgroup Server	This service manages workgroups and queues.
ShoreTel-Zin	ShoreWare Database Management Service	This service manages and updates the ShoreWare database.

Table 18-1 ShoreWare Service Areas

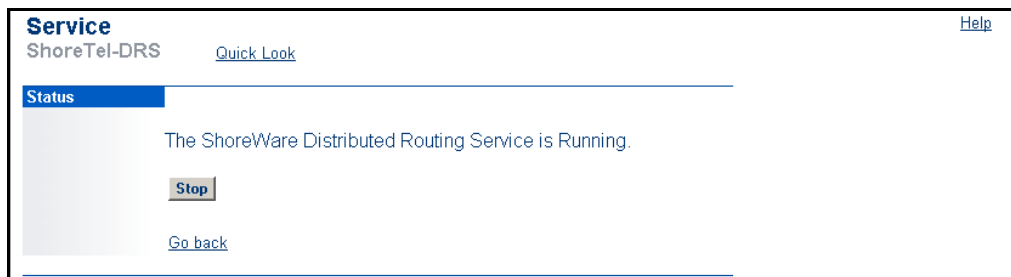


Figure 18-15 Service Maintenance Page

- **Informational:** The number of informational messages that were reported. Informational messages provide the status of a service, switch, or port.

18.3.10 Quick Look Server Maintenance—Distributed Servers

The *Quick Look Distributed Server Maintenance* page is shown in Figure 18-16. This page is hosted on a web site on the distributed server. The information provided is a subset of the main server maintenance page, since not all ShoreWare services run on the distributed server.

18.4 Switch Connectivity

The *Switch Connectivity* page (Figure 18-17) lists all ShoreGear voice switches when Distributed Routing Service (DRS) is disabled. When DRS is enabled, the switch connectivity table is organized by site.

The *Switch Connectivity* page includes the following:

- **Connectivity Grid:** The following indicators provide information about the listed switch:
 - Green (C) indicates that the switch is connected and communicating with other switches in the system.
 - Yellow (U) indicates that the switch connectivity is unknown because it cannot communicate with TMS.
 - Red (N) indicates that the switch has lost communications with the server.

Distributed Server Maintenance on 10.7.0.34 Last updated: 4/22/2004 12:26:36 PM (GMT-07:00) [refresh](#) [Help](#)
Local time: 4/22/2004 12:26:59 PM (GMT-07:00)

[Go to Quick Look Page](#)

Status

Name	Status
Communication	OK
TAPI	OK
Notification	OK
Software Version	OK
SMTP Send	OK

Services [Go to All Services](#)

Service	Description	Status	Command
ShoreTel-CSISSVC	ShoreWare CSIS Server	Running	Select...
ShoreTel-DRS	ShoreWare Distributed Routing Service	Running	Select...
ShoreTel-DTASSvc	ShoreWare Distributed Telephony Application Service	Running	Select...
ShoreTel-EventWatch	ShoreWare Event Watch Server	Running	Select...
ShoreTel-IPDS	ShoreWare IP-Phone Display Service	Running	Select...
ShoreTel-MailServ	ShoreWare Voice Mail Message Server	Running	Select...
ShoreTel-PortMgr	ShoreWare Voice Mail Port Manager	Running	Select...
ShoreTel-SoftSwitch	ShoreWare Software Telephony Switch	Running	Select...
ShoreTel-TMS	ShoreWare Telephony Management Server	Running	Select...
ShoreTel-Vmail	ShoreWare Voice Mail Application	Running	Select...

Events [Go to Application Log](#) [Go to System Log](#)

Date	Errors	Warnings	Informational
Today	-	-	-
Yesterday	-	-	2

Figure 18-16 Quick Look Server Maintenance—Distributed Server

Switch Connectivity on localhost Last updated: 8/13/2004 3:21:27 PM (GMT-07:00) [refresh](#) [Help](#)
Local time: 8/13/2004 3:22:04 PM (GMT-07:00)

[Go to Maintenance - Switches Summary page](#)

Connectivity of All Switches							
	1	2	3	4	5	6	7
Bedrock	1	N	U	U	U	U	U
Contact Center	2	N	C	C	N	C	N
Doc-24	3	N	C	C	N	C	N
First T1	4	U	U	U	N	U	U
ShoreDoc-HQ	5	N	C	C	N	C	N
Teleworker	6	U	U	U	U	U	N
ZerhGut	7	U	U	U	U	U	N

C Connected
N Not Connected
U Unknown because TMS cannot communicate with switch

Legend:

Hint: Place mouse over any cell on the connectivity grid, see detail in status bar at bottom left.

© 1998-2004 ShoreTel, Inc. All rights reserved.

Figure 18-17 Switch Connectivity Page

For more detailed information about a switch, place the mouse cursor over the connectivity cell in question and status information is displayed in the status bar at the bottom of your browser.

18.5 Conference Ports

The Conference Ports page shows the number of ports currently being used for conferencing and the number of free ports allocated for conferencing for each site in the system. The number of Make Me Conference calls active at each site is also shown. As with the Quick Look page, clicking on an entry in the Site column displays the *Maintenance Switches Summary* page described earlier in this chapter (refer to Figure 18-3).

Site	In-Use		Free		Percent
	Active Calls	Ports	Ports		
Headquarters	0	0	4		100%
El Dorado	0	0	0		0%
Remote-Site	0	0	0		0%
Bonn	0	0	0		0%
Total	0	0	4		100%

© 1998-2004 ShoreTel, Inc. All rights reserved.

Figure 18-18 Conference Ports Page

There are two main section displayed on the Conference Ports page. The first is data about in-use ports and the second is data about free ports.

- **In-Use**
 - *Active Calls*: Lists the number of calls active on the ports at a site
 - *Ports*: The number of ports that are in use at each site.
- **Free**
 - *Ports*: The number of ports that are free at each site.
 - *Percent*: The percentage of total ports at a site that are currently free.

18.6 Event Log

All events are reported to and viewed from the *System Event Log* and *Application Event Log* maintenance pages. To refresh these pages, click the *refresh* link. Clicking the *Quick Look* link takes you to the *Quick Look* maintenance page, described in Section 18.2 on page 346.

Figure 18-19 displays an Application Event Log.

Figure 18-20 displays a System Event Log.

ShoreTel 6.1 system events can also be viewed with the Windows Event Viewer.

Event categories for ShoreWare sources are reported to the applications log, in the following categories:

- Event Watch
- Java Client
- Java Server
- Notification

Application Event Log Last updated: 4/14/2004 3:06:15 PM (GMT -07:00) [refresh](#) [Help](#)
on localhost

[Go to Quick Look page](#)

Events (Date Format: mm/dd/yyyy) Previous Next

Date	Time	Source	Category	Event
4/14/2004	1:46:50 PM	ShoreWare	Switch	1305
4/14/2004	1:17:21 PM	ShoreWare	DRS	3108
4/14/2004	1:13:40 PM	ShoreWare	System Mgmt Interface	900
4/14/2004	12:33:28 PM	ShoreWare	DRS	3109
4/14/2004	12:29:07 PM	ShoreWare	TMS	238
4/14/2004	11:04:44 AM	ShoreWare	DRS	3109
4/14/2004	11:03:58 AM	Msiinstaller	None	1015
4/14/2004	11:03:58 AM	Msiinstaller	None	1001
4/14/2004	11:03:58 AM	Msiinstaller	None	1004
4/14/2004	10:38:02 AM	ShoreWare	DRS	3108
4/14/2004	10:30:04 AM	ShoreWare	System Mgmt Interface	900
4/14/2004	10:12:51 AM	ShoreWare	TMS	238
4/14/2004	10:12:51 AM	ShoreWare	TMS	238
4/14/2004	10:09:17 AM	Active Server Pages	None	3
4/14/2004	10:01:50 AM	ShoreWare	Switch	1305
4/14/2004	10:00:50 AM	ShoreWare	CSIS	2402
4/14/2004	9:47:11 AM	ShoreWare	Switch	115
4/14/2004	9:47:11 AM	ShoreWare	Switch	163
4/14/2004	9:47:11 AM	ShoreWare	Switch	1324
4/14/2004	9:47:11 AM	ShoreWare	TMS	234
4/14/2004	9:47:10 AM	ShoreWare	DRS	3109
4/14/2004	9:46:33 AM	ShoreWare	TMS	233
4/14/2004	9:46:16 AM	ShoreWare	Switch	115
4/14/2004	9:46:16 AM	ShoreWare	Switch	163
4/14/2004	9:46:16 AM	ShoreWare	Switch	107
4/14/2004	9:46:16 AM	ShoreWare	TMS	234
4/14/2004	9:46:00 AM	ShoreWare	DRS	3108
4/14/2004	9:45:38 AM	ShoreWare	TMS	233
4/14/2004	9:45:23 AM	ShoreWare	TMS	220
4/14/2004	9:45:23 AM	ShoreWare	TMS	234
4/14/2004	9:45:23 AM	ShoreWare	TMS	212
4/14/2004	9:45:23 AM	ShoreWare	TMS	233
4/14/2004	9:45:23 AM	ShoreWare	TMS	220
4/14/2004	9:45:23 AM	ShoreWare	TMS	203
4/14/2004	9:45:23 AM	ShoreWare	Switch	178
4/14/2004	9:45:04 AM	ShoreWare	Switch	177
4/14/2004	9:45:01 AM	ShoreWare	Switch	178
4/14/2004	9:44:51 AM	ShoreWare	DRS	3109

Figure 18-19 Application Event Log panel

- Port Mapper
- Software Telephony Switch
- Switch
- System Management Database
- System Management Interface
- Voice Mail Application
- Voice Mail Message Server
- Voice Mail Port Mapper
- Workgroup Server

18.6.1 Viewing Event Reports

To view an event report, click an entry in the *Date* column on the Services or Application Event Log page. The associated *Event Info* page is invoked as shown in Figure 18-21.

System Event Log Last updated: 5/18/2004 4:13:29 PM (GMT -07:00) [refresh](#) [Help](#)
on localhost

[Go to Quick Look page](#)

Events (Date Format: mm/dd/yyyy) [Previous](#) [Next](#)

Date	Time	Source	Category	Event
5/18/2004	3:17:30 PM	Schannel	None	36871
5/18/2004	3:17:30 PM	Schannel	None	36871
5/18/2004	3:16:12 PM	ShoreWare-Media	Media Driver	2100
5/18/2004	3:15:56 PM	EventLog	None	6005
5/18/2004	3:15:56 PM	EventLog	None	6009
5/18/2004	3:14:41 PM	EventLog	None	6006
5/18/2004	12:00:52 PM	Schannel	None	36871
5/18/2004	12:00:52 PM	Schannel	None	36871
5/18/2004	11:30:42 AM	ShoreWare-Media	Media Driver	2100
5/18/2004	11:30:33 AM	DCOM	None	10002
5/18/2004	11:30:28 AM	EventLog	None	6005
5/18/2004	11:30:28 AM	EventLog	None	6009
5/18/2004	11:29:05 AM	EventLog	None	6006
5/18/2004	11:28:36 AM	DCOM	None	10002
5/18/2004	11:28:05 AM	DCOM	None	10002
5/18/2004	11:27:35 AM	DCOM	None	10002
5/18/2004	11:27:05 AM	DCOM	None	10002
5/18/2004	11:26:35 AM	DCOM	None	10002
5/18/2004	11:26:05 AM	DCOM	None	10002
5/18/2004	11:25:35 AM	DCOM	None	10002
5/18/2004	11:25:05 AM	DCOM	None	10002
5/18/2004	11:24:35 AM	DCOM	None	10002
5/18/2004	11:24:05 AM	DCOM	None	10002
5/18/2004	11:23:32 AM	DCOM	None	10002
5/18/2004	11:23:02 AM	DCOM	None	10002
5/18/2004	11:22:32 AM	DCOM	None	10002
5/18/2004	11:22:02 AM	DCOM	None	10002
5/18/2004	11:21:32 AM	DCOM	None	10002
5/18/2004	11:21:02 AM	DCOM	None	10002
5/14/2004	1:56:28 AM	Schannel	None	36871
5/14/2004	1:56:28 AM	Schannel	None	36871
5/13/2004	8:10:13 PM	ShoreWare-Media	Media Driver	2100
5/13/2004	8:09:57 PM	EventLog	None	6005
5/13/2004	8:09:57 PM	EventLog	None	6009
5/13/2004	8:08:43 PM	EventLog	None	6006
5/13/2004	8:08:32 PM	MRxSmb	None	3034
5/13/2004	8:08:32 PM	MRxSmb	None	3034
5/13/2004	8:08:32 PM	BROWSER	None	8021

Figure 18-20 System Event Log Page

The *Event Info* page provides the following information, from which you can ascertain the severity of the event:

- Source reporting the event
- Category of the event
- Event number
- Event description
- Date and time
- Event type (error, warning, or information)

Clicking the *Go back* link takes you back to the top-level event log (System or Application) page.

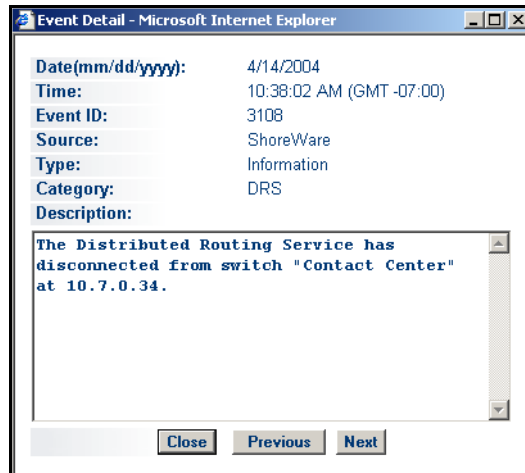


Figure 18-21 Event Info Page

18.7 Services

The *Services* maintenance page, shown in Figure 18-22, lists all the services that reside on the server. It includes the name of the service, its description, and its operational status (running, not running, or paused).

To refresh the page, click the *refresh* link. Clicking the *Quick Look* link takes you to the *Quick Look* maintenance page, described in Section 18.2 on page 346.

Green, upward-pointing arrows indicate that the service or application is running. Red, downward-pointing arrows indicate that the service or application is not running or paused.

18.7.1 Service Maintenance Page

To start a service that is not running or paused, click the service name to invoke the *Service Maintenance* page, shown in Figure 18-23.

From this page you can start or stop the service by clicking the *Start* or *Stop* button. Clicking the *Quick Look* link takes you back to the *Quick Look* maintenance page, described in Section 18.2 on page 346. Clicking the *Go back* link takes you back to the *Services* page.

18.8 Event Filters

Event filters specify the criteria for which e-mail notifications are sent after an event has been reported. The *Event Filters* list page, shown in Figure 18-24, displays a list of event filters that you create. You invoke this page by clicking *Event Filters* from under the *Maintenance* link in the navigation frame.

The *Event Filters* list page provides the following information:

- **Source:** The name of the event source.
- **Category:** The category for the source.
- **Event ID:** The event number that is assigned to the filter.
- **Type:** The event type that is assigned to the filter.
- **Server:** The server that the event filter runs on.

Services Last updated: 4/14/2004 3:10:35 PM (GMT -07:00) [refresh](#) [Help](#)
 on localhost Local time: 4/14/2004 3:10:44 PM (GMT -07:00)
[Go to Quick Look page](#)

Name	Description	Status
Alerter	Alerter	Running
AppMgmt	Application Management	Not Running
BITS	Background Intelligent Transfer Service	Running
Browser	Computer Browser	Running
cisvc	Indexing Service	Not Running
ClipSrv	ClipBook	Not Running
Dfs	Distributed File System	Running
Dhcp	DHCP Client	Running
dmadmin	Logical Disk Manager Administrative Service	Not Running
dmserver	Logical Disk Manager	Running
Dnscache	DNS Client	Running
Eventlog	Event Log	Running
EventSystem	COM+ Event System	Running
Fax	Fax Service	Not Running
IISADMIN	IIS Admin Service	Running
ismSrv	Intersite Messaging	Not Running
kdc	Kerberos Key Distribution Center	Not Running
lanmanserver	Server	Running
lanmanworkstation	Workstation	Running
LicenseService	License Logging Service	Running
LmHosts	TCP/IP NetBIOS Helper Service	Running
McAfeeFramework	McAfee Framework Service	Running
McShield	Network Associates McShield	Running
mesvc	Monitor Control Service	Not Running
McTaskManager	Network Associates Task Manager	Running
Messenger	Messenger	Running
nmmsrvc	NetMeeting Remote Desktop Sharing	Not Running
MSDTC	Distributed Transaction Coordinator	Running
MSFTPSVC	FTP Publishing Service	Running
MSIServer	Windows Installer	Not Running
NetDDE	Network DDE	Not Running
NetDDEdsdm	Network DDE DSDM	Not Running
Netlogon	Net Logon	Not Running
Netman	Network Connections	Running
NtFrs	File Replication	Not Running
NtLmSsp	NT LM Security Support Provider	Not Running
NtmsSvc	Removable Storage	Running

Figure 18-22 Services Page

Service [Quick Look](#) [Help](#)
 ShoreTel-DRS

Status

The ShoreWare Distributed Routing Service is Running.

[Go back](#)

Figure 18-23 Service Maintenance Page

- **Email:** The e-mail address that the filter reports to.

To edit an existing event filter, click its name in the *Source* column. This invokes the *Event Filter* edit page. Instructions for editing an existing filter and creating a new filter are provided in Section 18.8.1.

Source	Category	Event ID	Type	Server	Email
ShoreWare	CSIS	Any	Error	ShoreDoc-HQ	A.Ant@yourcompany.com
ShoreWare	Director Utilities	Any	Information	Contact Center	A.Ant@yourcompany.com

Figure 18-24 Event Filters List Page

18.8.1 Editing and Creating Event Filters

The *Event Filter* edit page, shown in Figure 18-25, lets you create new and edit existing event filters. To add a new event filter, click *add new* on the *Event Filters* list page, or click *New* or *Copy* on the *Event Filter* edit page.

Figure 18-25 Event Filter Edit Page

- **Parameters:** The parameters on the *Event Filter* edit are as follows:
- **Server:** This is the server that the event filter runs on. Click **All** or choose a server from the drop-down list.
- **Source:** These options define the event source for the filter. Select from the following:
 - *ShoreWare*—Notifies about ShoreWare events. The associated drop-down list lets you select the ShoreWare category. Selecting *Any* reports events about all ShoreWare categories.
 - *Services*—Notifies about any non-ShoreTel service.
 - *Other*—Notifies about any event source.
- **Category:** This lets you select a category when the *Services* or *Other* option is selected.
- **Event ID:** This lets you select the event identification number that is referenced in the e-mail notification.
- **Type:** These options let you select the severity type:
 - *Error*—Requires immediate attention.
 - *Warning*—Alerts to potential errors.

- *Information*—Provides status of service, switch, or port.
- *All*—Notifies about all three types.
- **Target E-mail Address:** This is the e-mail address of the intended system administration or technical support entity.

To create or edit an event filter:

- Step 1** If you are editing an existing filter, click the filter's name on the *Event Filters* page. To create a new filter, click *add new*.
- Step 2** Click *All* or choose a server from the *Server* drop-down list.
- Step 3** Select a source by clicking the *ShoreWare*, *Services*, or *Other* option.
Sources may have more than one filter associated with them, depending on how you set the criteria. For example, the ShoreTel Event Watch service might have two or more filters, each having different event types assigned to them.
- Step 4** Select a source and category based on the following:
- If the *ShoreWare* option is selected, select an event category from the *Category* drop-down list or select *Any* for all.
 - If *Services* is selected, select a service from the drop-down list, as well as a category.
 - If *Other* is selected, enter a service name, as well as a category.
- Step 5** Enter an event number from the *Event ID* drop-down list or select *Any*.
- Step 6** Select an event type by clicking the *Error*, *Warning*, *Information*, or *All* option.
- Step 7** Enter an e-mail address in the *Target E-mail Address* field.
- Step 8** Click *Save* to save the service's filter criteria.

18.9 Database Maintenance

ShoreWare uses MySQL to manage the ShoreWare configuration and CDR databases residing on the Headquarters server. MySQL databases support the UTF-8 character set. All ShoreWare configuration and CDR data is stored in UTF-8 format.

18.9.1 MySQL Services

18.9.1.1 Monitoring MySQL Service

ShoreWare reports MySQL service status on the Main Server Maintenance window. To access the Main Server Maintenance window, select *Maintenance* -> *Quick Look*, then perform one of the following:

- Click the first site on the list, typically named *Headquarters*, and click the name of the Soft Switch.
- Click the name of the Main Server, typically named *Headquarters Server*.

Figure 18-26 indicates the position of the MySQL service monitor on the Main Server Maintenance panel. To restart the MySQL service, access the MySQL Command drop down menu on the right side of the panel and select Start.

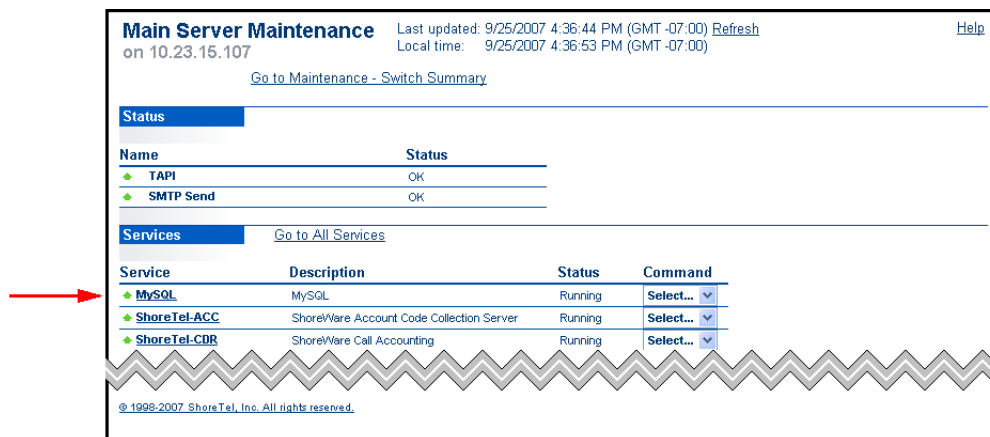


Figure 18-26 MySQL Service Monitor and Command – Main Server Maintenance panel

18.9.1.2 Browsing MySQL Database Tables

MySQL provides a query browser for viewing the database tables. The browser is available at <http://dev.mysql.com/downloads/gui-tools/5.0.html>

ShoreTel 6.1 does not support MySQL tools or any problems arise from using these tools.

18.9.1.3 Compatibility with Backup and Antivirus Utilities

Running a virus checker or a backup utility (disk or file) on a MySQL database file causes the MySQL service to crash.

When running these utilities, the MySQL database file (c:\Shoreline Data\Call Records 2\Data\[ibdata1, ib_logfile*]) and the folder C:\windows\temp must be included in the exclusion list.

18.9.2 MySQL Backup and Restore Utilities

MySQL provides utilities for backing up, restoring, and replicating MySQL databases. The following sections describe the processes for performing these functions.

18.9.2.1 Database Backup

The following command stores a backup copy of the ShoreWare configuration database:

```
(ShoreWare Path)\(Command Path)\mysqldump --add-drop-table --add-drop-database --routines --single-transaction --default-character-set=latin1 --databases db_name --user [username] --password [password] [database_to_dump] > [backupfile.sql]
```

where

- (ShoreWare Path) is the location where ShoreWare was installed
- (Command Path) is the location of mysqldump.exe in the installation directory
- backupfile.sql specifies the proposed location of the created backup file.
- db_name is either shoreware or shorewarecdr:
shoreware is entered to backup the configuration database
shorewarecdr is entered to backup the CDR database

Example: To backup the configuration file where:

- ShoreWare is installed at C:\Program Files\ShoreLine Communications\ShoreWare Server
- mysqldump.exe is located at C:\Program Files\ShoreLine Communications\ShoreWare Server\MySQL\MySQL Server 5.0\bin\
- the backup is to be placed at C:\shoreware_Bkup.sql
- the password is shorewaredba

Execute the following command line:

```
C:\Program Files\Shoreline Communications\Shoreware Server\MySQL\MySQL
Server 5.0\bin\mysqldump.exe --user=root --password=shorewaredba --
databases Shoreware --routines --single-
transaction>C:\shoreware_Bkup.sql
```

MySQL provides documentation on database backup at:

<http://dev.mysql.com/doc/refman/5.0/en/disaster-prevention.html>

18.9.2.2 Database Restore

The following command restores a backup copy of a ShoreWare configuration database as an active database:

```
(ShoreWare Path)\(Command Path)\mysql -u [username] -p [password]
[database_to_restore] < [backupfile.sql]
```

where

- *(ShoreWare Path)* is the location where ShoreWare was installed
- *(Command Path)* is the location of the mysqldump.exe within the installation directory
- *backupfile.sql* specifies the location of the backup file to be restored.

To restore a configuration file where:

- ShoreWare is installed at C:\Program Files\ShoreLine Communications\ShoreWare Server
- mysql.exe is located at C:\Program Files\ShoreLine Communications\ShoreWare Server\MySQL\MySQL Server 5.0\bin\
- the backup is located at C:\shoreware_Bkup.sql
- the password is shorewaredba

Execute the following command line:

```
"C:\Program Files\Shoreline Communications\Shoreware
Server\MySQL\MySQL Server 5.0\bin\mysql.exe" -u=root -p=shorewaredba
Shoreware < "c:\shoreware_Bkup.sql"
```

18.9.2.3 Database Replication

MySQL provides a built in database replication tool. Refer to the following for information on setting up the database replication:

http://www.howtoforge.com/mysql_database_replication

Refer to the following for tools, and add-ons that can assist in database replication:

<http://solutions.mysql.com/solutions/tools?type=25>

The database replication tool is provided by MySQL. ShoreTel 6.1 does not provide support for MySQL database replication.

System Recovery

19.1 Introduction

This chapter describes methods and tools that protect your system against computer failure. System recovery procedures provided or supported in this chapter include:

- **Backup:** Saves a copy of Headquarters, Distributed, or Voicemail Model Server files to a specified location.
- **Restore:** Restores ShoreWare server files previously saved by a Backup operation.

The Backup and Restore procedures in this chapter stop server operation, performs the backup or restore, then resumes server operation.

- **Failover:** Transfers control of a ShoreWare Server to a redundant computer.

Failover is the system function where, upon the failure or abnormal termination of the active server, system control is automatically transferred to a redundant or standby server, normally without human intervention or warning.

ShoreTel recommends Double-Take to provide failover protection for ShoreWare servers. For complete information on implementing Double-Take failover protection, refer to the Double-Take application notes on at the ShoreTel website.

19.2 Overview

19.2.1 Back Up

Backup saves a copy of server files which restores the server to a known state. ShoreTel provides script files that perform all required steps to back up Headquarters, Distributed, and Voicemail Model servers. Each server type requires a different backup script.

In addition to starting backup operations by running a script file from the Command Prompt, you can use MS Scheduler to initiate periodic, unattended backup operations.

Server activity must be stopped before backing up files to prevent corruption caused from modifying files during its backup. The procedures provided by ShoreTel stop the server before the backup and restarts the server after the backup is complete. ShoreTel recommends backing up files during scheduled down times or periods of light activity.

When backing up an entire system, ShoreTel recommends starting with the Distributed Servers. The Headquarters Server can remain operational while backing up files on one or more Distributed Servers. You can simultaneously back up multiple distributed servers. After the Distributed Servers are completed, then back up files on the Headquarters Server. This operational order minimizes the down time of the Headquarters server.

19.2.2 Restore

Restore returns a server to a previously known state by copying a set of server files previously created through the backup procedure to server directories. ShoreTel provides script and batch files that perform all necessary steps to restore Headquarters, Distributed, and Voicemail Model servers. Each server type requires different restore procedures.

You can restore all servers for which backup file exist or restore only selected servers. Operations and files received on a server after the backup was created are lost when files are restored to the server. When restoring a Headquarters server, all files on Distributed Servers that do not require restoring remain intact; however, voice mail received for mailboxes created since the backup was created may be lost regardless of the server upon which they reside.

Server activity must be stopped while files are restored to a server. The procedures provided by ShoreTel stops the server before the restoring the files and restarts it after the restoration is complete. Files must be restored to the directory direction from where they were stored.

When restoring an entire system, ShoreTel recommends restoring the Headquarters Server first, to establish a functioning system as soon as possible. After the Headquarters Server is restored, you can restore Distributed Servers while the Headquarters Server is active. This operational order minimizes the down time of the Headquarters server.

Files can only be restored to the server from which they were backed up. Backup files from the Headquarters server can restore only the Headquarters Server. For systems with more than one Distributed Server, backup files are not interchangeable between the servers.

19.3 Process Requirements

19.3.1 Protected Server Files

Backup and Restore operations protect the following ShoreWare Server folders and files:

- **Headquarters Server**
 - /inetpub/mailroot
 - /intepub/ftproot
 - /windows/my.ini
 - /shoreline data
 - /MessageFiles
 - /Prompts
 - /Scripts
 - /SoftSwitch
 - /Templates
 - /Call Records 2
 - /Database
 - /Logs
 - /Vms
- **Distributed Server**
 - /shoreline data
 - /MessageFiles
 - /Prompts
 - /SoftSwitch
 - /Templates
 - /Logs
 - /Vms

- Voicemail Model Server
 - /cf
 - cfg.dat
 - ShoreTel.cfg

19.3.2 Location of Script Files

ShoreTel provides Backup and Restore scripts and support files at:

`<SRoot>\Shoreline Communications\ShoreWare Server\Scripts\Sample_Backup_Restore`

where `<SRoot>` designates the folder where the ShoreWare Servers files were installed. In a normally installed system, `<SRoot>` is `C:\Program Files`

19.3.3 Backup File Location

Backup and Restore script files specify the directory to where the back up files are stored and from where they are retrieved. You can customize the storage location by modifying the script files. The default folder for Headquarters and Distributed backup files is:

`C:\Shoreware Backup\Backup.`

Voicemail Model server backup and restore procedures are initiated from a computer that

1. hosts the Headquarters server or a Distributed server
2. can establish an SSH connection to the Voicemail model server

Backup files for the Voicemail model server are stored on the computer that initiated the backup operation. The default location of these backup files is

`C:\Shoreware Backup\VMBackup\VMB_<ip address>`

where `<ip address>` is the Voicemail model server address. When a backup operation stores files from multiple VM Model servers, one `VMB_<ip address>` is created for each server.

Example: A backup procedure stores files from three Voicemail model servers, with IP addresses of 10.1.1.50, 10.1.1.52, and 10.1.1.60. After the back up is complete, the computer from where the procedure was initiated stores the server files in the following three directories:

`C:\Shoreware Backup\VMBackup\VMB_10.1.1.50`
`C:\Shoreware Backup\VMBackup\VMB_10.1.1.52`
`C:\Shoreware Backup\VMBackup\VMB_10.1.1.60`

Examples of backup file folders in this chapter were generated with the backup location set to the default value.

19.3.4 Main Scripts

The script files that initiate backup and restore operations, respectively, are

- shoreware_backup.wsf [*server_type*] [all | db | vm | log]
- shoreware_restore.wsf [*server_type*] [all | db | vm | log]

— *The first argument specifies the server type:*

hq – Headquarters server
 dvm – Distributed Voice Mail server
 vmb – Voicemail Model server

You must designate one server type.

— *The second argument specifies the server component*

db – database (not valid if the server_type = dvm or vmb)
 vm – voicemail (not valid if server type = vmb)
 log – log files (not valid if server type = vmb)
 all – database, voicemail, and log files

You must designate at least one value.

— *Switches*

/v – verbose mode

19.3.5 Convenience Scripts

ShoreTel provides the following batch files that backup the complete set of server files for the specified server type:

- hq_backup_all.bat
- dvm_backup_all.bat
- vmb_backup_all.bat

ShoreTel provides the following batch files that restores the complete set of server files for the specified server type:

- hq_restore_all.bat
- dvm_restore_all.bat
- vmb_restore_all.bat

19.3.6 Support Files

ShoreTel provides the following files that support the core backup and restore scripts and batch files:

- common.vbs – contains the VBS library file for common code
- plink.exe – a command line tool for executing commands on a remote machine
- pscp.exe – a tool that transfers files securely through an SSH connection
- sw_backup_restore.ini – configuration file for backup and restore

19.3.7 Estimated Completion Time

The following is an estimate of time required by ShoreWare Backup procedures:

- **Clean System (No VM or CDR)**
 - Total – 379 secs
 - Backup VM – 2 secs
 - Backup CDR– 35 secs
- **Loaded System (VM: 100 messages/13.5 MB – CDR: 500,000 calls)**
 - Total – 508 secs
 - Restore VM – 21 secs
 - Restore CDR – 104 secs

The following is an estimate of the time required to Restore ShoreWare Server files:

- **Clean System**
 - Total – 416 secs
 - Backup VM – 3 secs
 - Backup CDR – 32 secs
- **Loaded System (VM: 100 messages/13.5 MB – CDR: 500,000 calls)**
 - Total – 525 secs

- Restore VM – 22 secs
- Restore CDR – 98 secs

19.3.8 Modifications for Non-Standard Installations

The scripts perform backup and restore operations for computers where the operating system and ShoreWare are installed on the C: drive. Backup and restore script for other configurations require adjustments to a registry setting.

To configure a system to backup and restore files on a non-standard configuration, perform the following:

Step 1 Access `sw_backup_restore.ini`

Step 2 Set `Windows.Install.Drive` to the drive where the OS is installed

Example: If Windows is installed on D drive, set `Windows.Install.Drive = C:`

Step 3 Point `Shoreware.Scripts.Root.Directory` to the drive where ShoreWare is installed.

Step 4 Point `PLINK.CMD` and `PSCP.CMD` to the drive where ShoreWare is installed.

19.4 Procedure

The following sections list the batch files that comprise the backup procedures for the Headquarters server, Distributed server, and Voicemail Model server. These files store the backup files to a location specified by `sw_backup_restore.ini`; storing files to a different backup location requires a change to this .ini file. The computer registry specifies the installation folder of the ShoreWare system and data files; the Backup procedure references this file to determine the location of the files being backed up.

19.4.1 Setting Up an Alternate Storage Folder

The following procedures stores the backup data in the following file directory:

`C:\Shoreware Backup\Backup\Shoreline Data`

This location is specified by the following lines in `sw_backup_restore.ini`:

```
[Backup Options]
Backup.Drive = C:
Backup.Root.Directory = \Shoreware Backup\Backup
Backup.ShoreWare.Directory = Shoreline Data\
Backup.Archive.Size = 2 ; folders
```

To store the backup files to a different drive, change the second line:

```
Backup.Drive = C:
```

To store the backup files to a different directory, change the third and fourth lines:

```
Backup.Root.Directory = \Shoreware Backup\Backup
Backup.ShoreWare.Directory = Shoreline Data\
```

19.4.2 Establishing a Connection to a Voicemail Model Server

VM Model servers Backup and Restore operations are performed from a Headquarters or Distributed Server by establishing an SSH connection between the servers. Before backing up or restoring Voicemail Model server files, the following data must be added to system files on the server from where the operations are initiated:

- RSA key of the VM Model server must be cached to the server's registry
- VM Model server address to the VMB.ip.list in the server's sw_backup_restore.ini file.

To cache the RSA key of the Voicemail Model to a server's registry:

Step 1 Open the Command prompt on the server

Step 2 Run `plink <ip address>` in the command prompt, where `<ip address>` is the ip address of the Voicemail Model server.

`plink.exe` is available on the internet; ShoreWare includes `plink.exe`, in the default configuration at the following location

`C:\Shoreline Data\Database\Scripts\Sample_Backup_Restore`.

The program returns a message indicating the storage status of the RSA key in the server's registry. Figure 19-1 displays the messages if the key is not present.

Step 3 If the key is not cached to the registry, press `y` when prompted.

Figure 19-1 displays the messages prior to and after the RSA is cached.

```
C:\Shoreline Data\Database\Scripts\Sample_Backup_Restore.>plink 10.1.1.242
The server's host key is not cached in the registry. You have no guarantee that the server is
the computer you think it is.
The server's rsa2 key fingerprint is: ssh-rsa 2048
08:b0:43:35:fd:21:1b:c2:6b:27:b4:3f:a9:f7:be:2d
If you trust this host, enter "y" to add the key to PuTTY's cache and carry on connecting.
If you want to carry on connecting just once, without adding the key to the cache, enter "n".
If you do not trust this host, press Return to abandon the connection.
Store key in cache? (y/n) y
login as: ^C
```

Figure 19-1 Caching the registry key by using `plink`.

To store a VM Model server address to the VMB.ip.list in the server's `sw_backup_restore.ini` file:

Step 1 Use Notepad or another text editor to open `sw_backup_restore.ini`

Step 2 Add the following code block to the opened file:

```
[VMB]
VMB.user = root
VMB.password = ShoreTel
VMB-Backup.Folder = VMB
VMB.ip.list = <ip address 1>, <ip address 2>, ... , <ip address n>
```

where `<ip address x>` represents the ip address of the Voicemail Model server whose files will be saved or restored from the server.

Example: Assume a server is to save or restore files from three VM Model servers, located at 10.1.1.24, 10.1.1.25, and 10.1.1.26. Then the VMB.ip.list line should be:

VMB.ip.list = 10.1.1.24, 10.1.1.25, 10.1.1.26

Step 3 Save and close the file.

19.4.3 Backing up Server Files

19.4.3.1 Manual Backup

To back up ShoreWare server files, enter one of the following file names in a Command Prompt window, as shown in Figure 19-2.

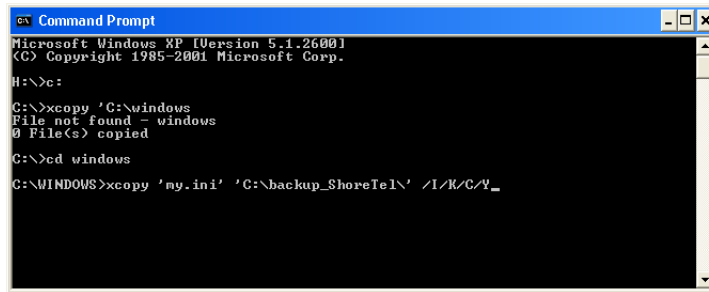


Figure 19-2 Command Prompt window

To perform a complete backup of a server, enter one of the following:

- hq_backup_all.bat
- dvm_backup_all.bat
- vmb_backup_all.bat

Routines for a Headquarters and Distributed server are initiated from the computer hosting the server. The Voicemail Model server routine is initiated from a computer configured as described in Section 19.4.2.

To backup a partial server, enter the following with the options listed in Section 19.3.4:

- cscript shoreware_backup.wsf [*server_type*] [all | db | vm | log]

Examples:

- Back up all Headquarters Server files – cscript shoreware_backup.wsf hq all
- Back up Headquarters database and VM files – cscript shoreware_backup.wsf hq db vm
- Back up Distributed log and voicemail files – cscript shoreware_backup.wsf dvm db log
- Back up all Voicemail Model Server files – cscript shoreware_backup.wsf vmb all

19.4.3.2 Scheduled Backup

The following procedure uses Microsoft Scheduler to schedule periodic backup operations:

Step 1 Open the Scheduled Tasks window, as shown in Figure 19-3.

Open this window by pressing the Start button and selecting *Programs -> Accessories -> System Tools -> Scheduled Tasks*.

Step 2 Open the Scheduled Task Wizard by double-clicking the Add Schedules Task icon in the Scheduled Tasks window.

Step 3 Step through the first two Scheduled Task Wizard panels, shown in Figure 19-4:

Panel 1: Press the Next button

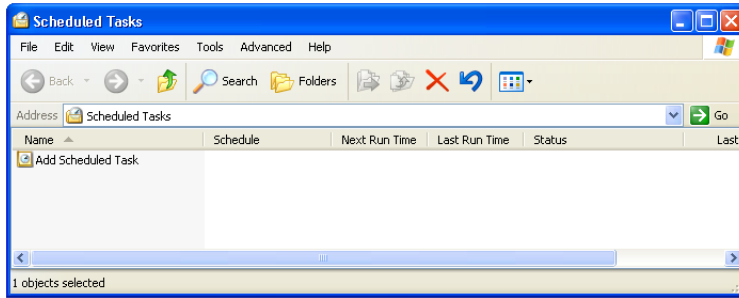


Figure 19-3 Scheduled Tasks Windows

Panel 2: Specify the file to be executed by pressing the Browse button, navigating to the folder containing the backup batch file.

In the default ShoreWare installation, the batch file is located in the following folder: *C:\Shoreline Data\Database\Scripts\Sample_Backup_Restore*

Panel 1



Panel 2

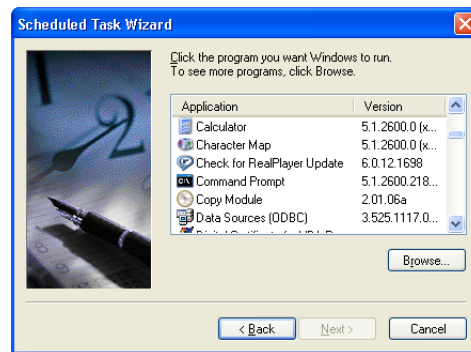


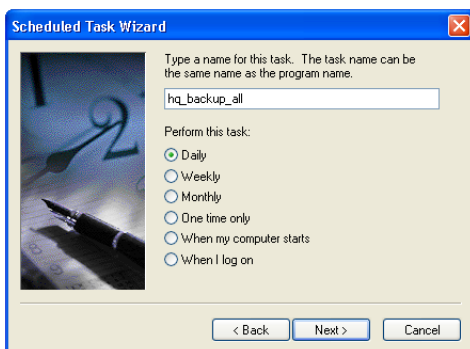
Figure 19-4 Scheduled Task Wizard – Panels 1 - 2

Step 4 Step through the next two Scheduled Task Wizard panels, shown in Figure 19-5.

Panel 3: Verify the correct file name is listed in the data entry field, then select the desired execution frequency.

Panel 4: Select the desired time, frequency, and start date.

Panel 3



Panel 4



Figure 19-5 Scheduled Task Wizard – Panels 3 - 4

Step 5 Step through the last two Scheduled Task Wizard panels, shown in Figure 19-6.

Panel 5



Panel 6



Figure 19-6 Scheduled Task Wizard – Panels 5 - 6

Panel 5: If necessary, enter the user name and password for the user running the batch file.

Panel 6: Verify the backup parameters are correct, then press the Finish Button to display the Scheduled Task window, shown in Figure 19-7.

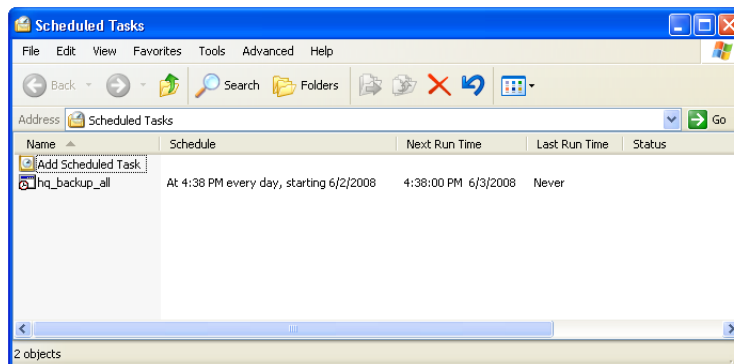


Figure 19-7 Scheduled Tasks window with Backup Task

Step 6 Verify the Scheduled Tasks window displays the Backup file name, then close the window by select *File -> Close* from the main menu.

19.4.4 Restoring a Server

To back up ShoreWare server files, enter one of the following file names in a Command Prompt window, as shown in Figure 19-2.

To restore all files that were stored in the most backup, enter one of the following:

- hq_restore_all.bat
- dvm_restore_all.bat
- vmb_restore_all.bat

The Headquarters and Distributed server routines are initiated from the computer that hosts the servers. The Voicemail Model server routine is initiated from a computer that has been configured as described in Section 19.4.2.

To perform a partial restoration of a server, enter the following, using options described in Section 19.3.4:

- cscript shoreware_restore.wsf [server_type] [all | db | vm | log]

Examples:

- Restore all available Headquarters Server files – cscript shoreware_restore.wsf hq all
- Restore Headquarters database and voicemail files – cscript shoreware_restore.wsf hq db vm
- Restore Distributed log and voicemail files – cscript shoreware_restore.wsf dvm db log
- Restore all Voicemail Model Server files – cscript shoreware_restore.wsf vmb all

19.4.5 Log Files

Log files display the commands that are performed during Backup and Restore operations at SwBackupRestore.log. By default, Windows maintains three log files.

Reporting

This chapter provides information about the reporting capabilities of the ShoreTel 6.1 system. The sections in this chapter are:

- Overview
- Reports
- Options

20.1 Overview

The ShoreTel 6.1 system provides reporting information through ShoreWare Director, so you can generate Call Detail Record (CDR) reports via Director, or you can modify the reporting options. The information is displayed on the reporting pages that you access from the *Reporting* link in the navigation frame, as shown in Figure 20-1.

Figure 20-1 Reports Page

Reporting pages include:

- Reports
- Options

20.2 Reports

Director is used to generate web-based Call Detail Record (CDR) reports from a local host or a remote server. Generating Reports from a remote server requires a Web Report License.

To generate reports from a local host:

Step 1 Log into director by entering `http://localhost/ShoreWareDirector/login.asp`

Step 2 Access the Reports page by selecting Reporting -> Reports in the menu panel

To generate reports from a remote site:

Step 1 Log into director by entering `http://<remote_address>/ShoreWareDirector/login.asp`, where `<remote_address>` is the IP address of the system you are accessing.

Step 2 Access the Reports page by selecting Reporting -> Reports in the menu panel

A Web Report License is required to access this page remotely.

The **Reports** page, shown in Figure 20-2, generates web-based Call Detail Records (CDR) reports. This page is the primary method of accessing and viewing the CDR data in the MySQL database.

Reports can be run from ShoreWare Director. After the report has been generated, it can be printed, exported, and navigated interactively, similar to compiled reports. By purchasing the proper keyed license (i.e. the Remote Web Reporting License) users can run a web-based report remotely from clients other than the headquarters machine (see Section 2.9 on page 25 for more information on obtaining the proper license).

Details:

- No more than two users should run reports at the same time. Having more than two people generating reports simultaneously could adversely impact system performance.
- The content and layout of all existing reports are essentially unchanged relative to the compiled reports that existed in versions prior to ShoreTel 7, except for the following:
 - Enhancements associated with WG overflow
 - Ability to report on internal calls for User reports
 - Ability to report on LAN calls for Media Stream reports
- The ability to generate web-based CDR reports is part of the ShoreTel 6.1 offering and is associated with the changeover from the Microsoft JET database management system to the MySQL database management system. See Section 20.2.3 on page 386 for information on porting over existing CDR data to the new MySQL format if you are upgrading to ShoreTel 7 (or higher) from a prior release, and if you want this old CDR data to appear in the web-based reports.

20.2.1 Reports

- **Please Select the Database:** This drop-down menu lists the active and archive databases from which you can generate reports.
- **Please select the report you want to run:** This drop-down lists all of the different types of reports that can be run. Options are:

ShoreTel Reports

Reports

Please select the Database: Main

Please select the report you want to run: Account Detail Report Run Report

Parameters

Enter the Account code(s): for all Account codes:

Selected Values: Account Detail Report
Account Summary Report
WAN Media Stream Detail Report
WAN Media Stream Summary Report
Trunk Activity Detail Report
Trunk Activity Summary Report
Workgroup Queue Summary Report
Workgroup Service Level Summary Report
User Activity Detail Report
User Activity Summary Report
Workgroup Agent Detail Report
Workgroup Agent Summary Report

Enter the extension, or range of extensions, you want to report on. You can add multiple extensions, or you can leave this field blank to report on all extensions:

Start of Range
 End of Range Add
 No lower value No upper value

Selected Values:

Enter the date range you want to report on:

Start of Range 01/08/2007
 End of Range 01/08/2007
 No lower value No upper value

Enter the time range you want to report on:

Start of Range 12 00 AM
 End of Range 11 59 PM
 No lower value No upper value

© 1998-2007 ShoreTel, Inc. All rights reserved.

Figure 20-2 Reports Page

- Account Detail Report
- Account Summary Report
- Media Stream Detail Report
- Media Stream Summary Report
- Trunk Activity Detail Report
- Trunk Activity Summary Report
- Workgroup Queue Summary Report
- Workgroup Service Level Summary Report
- User Activity Detail Report
- User Activity Summary Report
- Workgroup Agent Detail Report
- Workgroup Agent Summary Report

Refer to Section C.3 on page 425 for more information.

20.2.2 Parameters

- **Enter the account code(s) you want to report on:** Account codes are typically used to assist ShoreTel users in the billing of their clients (eg. a law firm tracking the length of a call). Enter the account code(s) that corresponds to one or more clients in the field provided and click the Add button to enter the number.
- **Enter the extension, or range of extensions, you want to report on:** Enter a single extension or a range of extensions in the fields provided to generate a report on

ShoreTel 8.1

385

only those extensions. You can also leave the fields blank to run the report on all extensions. You can enter an upper or lower extension and then select the **No lower value** or the **No upper value** radio button and to run a report on all extensions up to a specified extension.

- **Enter the date range you want to report on:** Click on the Calendar icon to select a **Start of Range** date and/or a **End of Range** date in the fields provided to generate a report on only those extensions.

Select the **No lower value** radio button and the **No upper value** radio button to run a report on run a report on all valid dates for which the system has CDR data.

- **Enter the time range you want to report on:** Click on the **Start of Range** radio button and use the drop-down menus to select the start time for the report. Click the **End of Range** radio button and use the drop-down menus to select the outside time limit for the report.

Select the **No lower value** radio button and the **No upper value** radio button to run a report on run a report on all hours within the specified dates.

20.2.3 Converting Existing CDR Data to MySQL Format

Releases prior to ShoreTel 7 relied on the Microsoft Joint Engine Technology (JET) database management system to store Call Detail Records (CDR) and to manipulate the data within the database. This approach limited the size of the database to 2 gigabytes, thus potentially limiting the scalability of the ShoreTel 6.1 system.

ShoreTel 7 replaced the Microsoft JET database management system with the more scalable MySQL technology. The maximum size for MySQL tables under Windows tables is 2 terabytes (TB), and the maximum size for the entire database is 64TB.

The Microsoft JET database management system automatically created a new database at the beginning of each month, and these database files could be viewed by simply double-clicking on them. In contrast, the MySQL technology does not partition the CDR data into separate files. Instead, all of the CDR data is stored in one large file, which can be viewed using a new Web-based Reporting feature via ShoreWare Director.

Upon installing ShoreTel 7, CDR data is stored in the MySQL database rather than the Microsoft Access database that was previously used by Microsoft JET. However, if a system is being upgraded to ShoreTel 7 (or higher) from a prior release, the original database will be retained on that system. New information will go into the MySQL database, and old information will remain in the old format.

When a system has been upgraded and a CDR report is generated on that system, only the MySQL database will be tapped to generate the report. Data in the old Microsoft Access database will be ignored when the report is generated. The old data will not appear in the report unless it has been converted to the new MySQL format using the command line utilities shown in the procedure below.

To import CDR data from a Microsoft Access database into MySQL:

Step 1 Open a command prompt window

Step 2 Change the command prompt directory to *C:\Program Files\Shoreline Communications \ShoreWare Server*

Step 3 Run the following command: `ImportCDR -s server -d databasename -f mdb-file`

- **server** is IP address of the mysql shorewarecdr database.
- **databasename** is shorewarecdr.
- **mdb-file** is the name of the mdb file (e.g. CDR.mdb).

The parameters are optional.

Details:

- By default, data is imported into the main CDR database on localhost.
- The main CDR.mdb is imported before any of the Archive mdb.
- Errors are written to error log, MakeCDRMMDDYYYY.log. The log will be present in Shoreline Data\Logs.

20.3 Options

The *Reporting Options* page (shown in Figure 20-3) is accessed by clicking on the Options link under Reporting.

The ShoreTel 6.1 system supports the ability to send CDR data out a serial port on the main ShoreWare server. The Reporting Options page allows a system administrator to designate which COM port to enable. CDR data is subsequently sent out this port (in addition to being sent to the regular text file and/or a database). Sending the CDR data out the serial port does not change the formatting.

Figure 20-3 Reporting Options Page

The Reporting Options Page is described as follows:

- **Refresh this page:** Click this link to update the information displayed on the page. The Reporting Options page automatically updates every 60 seconds.
- **COM Port for CDR Output:** Select the desired COM port on the headquarters server when CDR data will be sent. The default value is none, with a valid range of 1 – 10.
- **Retention Period for CDR Archive:** This value ranges from 1 day to 2000 days. Default value is 125 days.
- **Retention Period for CDR Data:** This value ranges from 1 day to 2000 days. Default value is 125 days.
- **Enable Archiving:** Select this check box to enable archiving.

- **Archive Database Name:** Enter the name of the archive database. Saving the name of the database will not create the archive database; a separate utility must be used. See Section 20.3.2 on page 389 for details on creating an archive database.
- **Archive Database IP Address:** Enter the IP address of the server where the archive database is stored.

20.3.1 Configuring Send CDR Out SMDR Interface

The ShoreTel 6.1 system captures Call Detail Records (CDR) in a database in a text file format. However, for legacy call accounting systems that cannot read CDR from a database, the CDR data can also be delivered via a Station Messaging Detail Record (SMDR) serial port on the main ShoreWare server by designating a COM port.

Details:

- Formatting of the CDR data remains the same, regardless of whether it is sent out the COM port or written to the database.
- The application should auto-detect the serial port configuration by extracting information about the status of the serial port configuration (e.g. baud rate) from the Windows registry.
- The feature will be disabled by default and must be enabled by selecting a COM port.
- If the serial port should become unavailable through an event such as becoming locked by extremely high volumes of traffic, the CDR data will be queued in a buffer for 300 seconds to help prevent the loss of data. If the serial port returns to service within the 300-second time period, the streaming will resume.

To designate a COM port in order to Send CDR Out SMDR Interface via ShoreWare Director:

Step 1 From the Reporting page, click on the Options link.

Step 2 Click on the **COM Port for CDR Output** drop-down menu and select the desired port on the headquarters server, as shown below:

The screenshot shows the 'Reporting Options' configuration page. At the top, there are 'Save' and 'Reset' buttons, and a 'Help' link. Below that, there is an 'Edit' link and a '* modified' indicator. The main configuration area includes a 'COM Port for CDR Output' dropdown menu with a list of options: 1, 2, 3, 4, 5, 6, 7, 8, 9, 10. The '2' option is selected. Other fields include 'Retention Period for CDR Archive' (1), 'Retention Period for CDR Data' (2), 'Enable Archiving' (unchecked), 'Archive Database Name', and 'Archive Database IP Address'.

Figure 20-4 Selecting the desired COM port

Step 3 Click **Save** to store your changes. Once the COM port configurations have been saved, the system will begin directing CDR data to the SMDR port as calls move through the system.

20.3.2 Creating the Archive Database

The following procedure describes the process of creating a separate (optional) archive database using the *MakeCDRArchive.exe* command line utility.

Step 1 To run this utility, verify that the following required files are located in the same directory:

- MakeCDR.dll
- MakeCDR.sql
- MakeCDR_sp.sql

By default, the installer puts them at the following location on the headquarters server:

C:\Program Files\Shoreline Communications \ShoreWare Server

Step 2 Open the command prompt window in the directory shown above and run the following command:

`MakeCDRArchive -s localhost -d databasename`

- databasename is the name of the archive database to be created. The database name must be the same as the name created on the **Options** page.
- If no name is defined, the default name of *shorewarecdrarchive* will be created within the following directory:
C:\Shoreline Data \Call Records2 \Data

Records from main database will be archived to the archive database when the services start up and every night at approximately 12 a.m.

Using Office Anywhere

Office Anywhere is a feature of the ShoreTel 6.1 system that allows a user to quickly and easily re-assign his or her extension to any telephone on the system or off the system. The user's communications profile is reassigned to that telephone, and calls placed to the user will be routed to that telephone, while calls placed from that telephone will reflect proper caller ID information.

The Office Anywhere feature might be used by the following types of users:

- Multi-site users, such as executives or managers, who might use the system across multiple locations. Office Anywhere allows these users to pick up a telephone at any location on the enterprise network, log into voice mail, and assign their extension to that telephone. (See Section A.5 on page 397.)
- Office hotel users, such as contractors or telecommuters, who may occasionally be out of the office or who might share a cubicle (and thus a phone) with another employee. Office Anywhere allows these users to have their own extension and mailbox, yet not have a dedicated switch port. They can simply assign their extension to a telephone on the network when they are in the office while allowing another user to usurp that phone when they are done with it. (See Section A.5 on page 397.)
- Remote or mobile users, such as employees in sales or support, who might travel frequently and would like to have all calls directed to their cell phone or home office PSTN phone. Office Anywhere allows these users to have their own extension and mailbox, yet not have a dedicated switch port, thus optimizing system resources. (See Section A.4 on page 395.)
- Legacy PBX Users, such as users with Off-System Extensions working with PCM. (See Section A.4 on page 395.)

Office Anywhere also allows the system administrator to configure all telephones as anonymous telephones and all users as virtual users, eliminating administrative costs associated with frequent moves. When a move occurs, users simply assign their extension to the telephone at the new location.

The feature also allows an off network user to manage a call via PCM, so while the conversation occurs over the cell phone or home phone, the call appears via PCM and can be controlled using many of the features available via PCM. Note that this requires the user to be located near a PC that has access to a broadband connection and that PCM is running.

Other benefits to the (Off-Net) Office Anywhere user include:

- Use the existing PSTN line for voice while managing the call via PCM over an ordinary broadband Internet connection
- Emulate analog extension hook switch actions via star-star (**) for FLASH and pound-pound (##) for on-hook/off-hook

- Access the user's directory numbers at the office
- The user appears to be calling from the office
- Keep communication costs minimized with flexible IP & trunking requirements
- Retain call management features of the ShoreTel 6.1 system over a broadband connection while maintaining audio quality over PSTN.

A.1 Special Considerations

The following considerations apply when using Office Anywhere:

- When an Office Anywhere call is completed and the carrier has not reported back to the ShoreTel 6.1 system that the far side has disconnected, instead of the call remaining active for a very long period of time, it will now remain active for approximately 5 seconds, after which it will disconnect. To initiate a new call during the 5-second window, you can press “##”.
- To use the Office Anywhere feature, the user must be in a user group that has a Telephony Class of Service of *Allow Extension Reassignment*. See Section A.3 on page 393.
- SIP is not supported as the external number to which the user is reassigned.
- ShoreTel 6.1 supports up to 1,000 virtual users.
- Incoming calls to the user's extension assigned to an off-net location will ring on their cell phone or Off-System Extension. If the call is not answered, normal call handling allows the caller to leave a message in the user's ShoreTel mailbox.
- Office Anywhere, when assigned to an off-net location, is fully controllable using PCM, except for answering a call, which must be done manually. The feature will also have limited TUI functionality.
- Office Anywhere calls that are terminated by hanging up using PCM do not produce standard dial tone. Office Anywhere uses a unique set of internal and external dial tones. This difference in tones may be important in certain installations if network devices have been configured to listen for normal class progress tones before taking action on a call, such as hanging up.
- Calls placed or answered using Office Anywhere, when assigned to an off-net location, will continue to show in the PCM call stack. Normal call control functions such as hold, unhold, conference, transfer, and park will continue to work. Park to the Office Anywhere extension, when assigned to an off-net location, is not supported.
- Office Anywhere, when assigned to an off-net location, will function somewhat like an automated "Find Me" feature, except the caller will not have to "press 1 to find me" because the PSTN phone number will be called immediately. The call can be answered by lifting the handset and pressing the DTMF digit 1 in response to the repeated prompt.
- Prompts (such as those used for Find Me) may be used to confirm answering. Answer style can be configured to one of two possibilities:
 - Wait for DTMF - The call is not put through until the user presses DTMF '1' (This is the default setting.)
 - Wait for Answer - The call is put through as soon as far-end answer is detected. (Note that far-end answer detection is not supported by the central office for

some trunk types, in which case the user must press '1'). (DTMF will be audible to the caller.)

A.2 Terminology

The terms used to describe Office Anywhere are as follows:

- **Anonymous telephone:** A telephone not currently assigned a user. You can make a call from an anonymous telephone, but you cannot call an anonymous telephone.
- **Any IP Phone:** The feature that lets users assign their extension to any IP phone on the enterprise network.
- **Assign:** The command that assigns an extension to a telephone.
- **Assigned:** The status of a user who is currently assigned to a telephone that is not their home phone.
- **Current telephone:** The telephone to which the user is currently assigned (also known as the current switch port).
- **Go Home:** The command to assign a user's extension back to his or her home telephone.
- **Home:** The status of a user who is assigned to his or her home telephone.
- **Home telephone:** The telephone to which the user is normally assigned (also known as the home switch port). This is the telephone to which the user returns when using the *Go Home* command.
- **Office Anywhere:** The feature that lets users assign their extension to any telephone, on-system or off-system extension.
- **Unassign:** The command that unassigns an extension from a telephone.
- **Vacated telephone:** A home telephone that currently does not have a user assigned. These are listed on the *Anonymous Telephones* edit page under *Vacated Telephones*.
- **Virtual user:** A user who does not have a physical telephone port and is currently assigned to the SoftSwitch.

A.3 Configuring Permission Settings

To configure the Office Anywhere feature, the system administrator must first give the end user permission to use the feature by following 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 **User Groups** link.
- Step 4** Create a new user group for the Office Anywhere users, or click on the name of an existing user group (in which all members will have access to Office Anywhere).
- Step 5** For the **COS - Telephony** field, click on the **Go to this Class of Service** link to display the following window:

Class of Service
Edit Telephony Features Permissions

[New](#) [Copy](#) [Save](#) [Delete](#) [Reset](#) [Help](#)

Edit this record [Refresh this page](#)

Name:

Max. Call Stack Depth:

Max. Parties in Make Me Conference:

Allow Call Pickup

Allow Trunk-to-Trunk Transfer

Allow Overhead and Group Paging

Allow Make Hunt Group Busy

Allow Extension Reassignment

Allow PSTN Failover

Show Caller ID Name and Number on Monitored Extensions

Allow Customization of IP Phone Buttons

Show Extensions with Different Prefixes in Directory

Allow Collaboration Features

Allow Recording of Own Calls

Directed Intercom / Paging: Allow Initiation
Accept: None All Only From: [Search](#)

Barge In: Allow Initiation
Accept: None All Only From: [Search](#)

Record Other's Calls: Allow Initiation
Accept: None All Only From: [Search](#)

Monitor Other's Calls: Allow Initiation
Accept: None All Only From: [Search](#)

Allow Call Handling Changes: Current Mode Detailed Settings

Allow External Call Forwarding and Find Me Destinations

Allow External Extension Reassignment

Scope:

Local Only

National Long Distance

International Long Distance

All Calls

Restrictions: (e.g. +1900,+1xxx976)

Permissions: (e.g. 1/311)

Figure A-1 Configuring the Edit Telephony COS window for Office Anywhere

Step 6 Select the **Allow Extension Reassignment** check box.

Step 7 Select the **Allow External Call Forwarding and Find Me Destinations** check box and the appropriate **Scope** radio button.

Step 8 Click **Save** to store your changes.

Details:

- If you are intending for a user to have access to the Office Anywhere feature, you should verify that he or she belongs to a User Group that is associated with the Class of Service you just modified above.

A.4 Configuring Off-Net External Assignment

The off-net external assignment is intended for mobile users who are not always in the office. These employees may travel frequently or may work from home, and they could benefit from having all calls directed to an “off-net” device, such as their cell phone or home office PSTN phone. Office Anywhere allows these users to have their own ShoreTel extension and mailbox without requiring a dedicated switch port and physical telephone in the office.

To assign your extension to an off-net location, follow the procedure below:

Step 1 Launch Personal Call Manager (PCM) on the client machine.

Step 2 Click on the File menu to display and select External Assignment, as shown below.

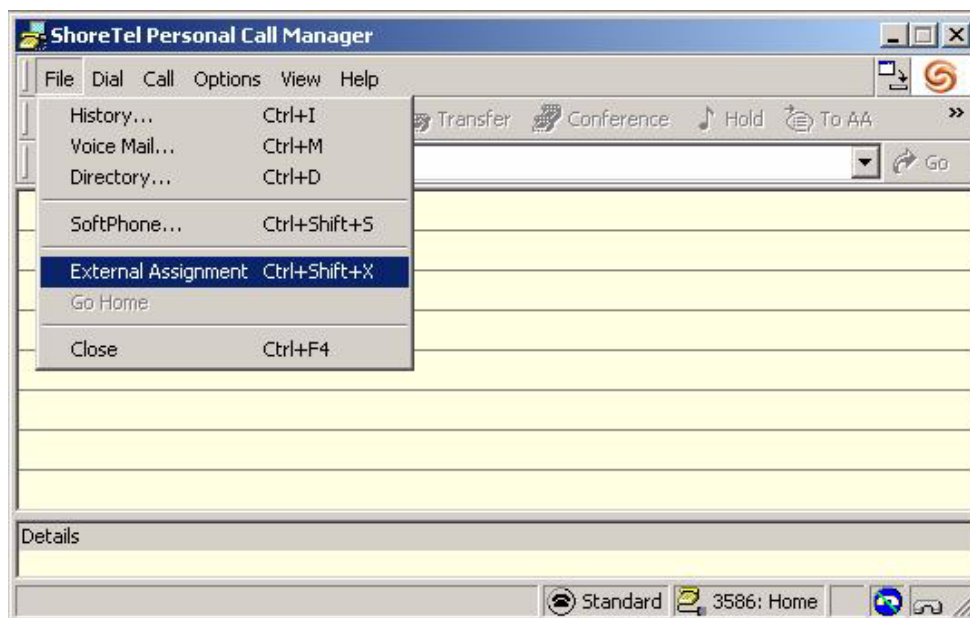


Figure A-2 Using PCM to configure the client machine for Office Anywhere

Step 3 When you select External Assignment, a pop-up window similar to the one shown below appears.

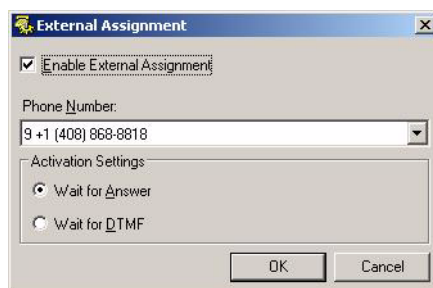


Figure A-3 Select the Enable External Assignment check box

Step 4 Select the Enable External Assignment check box

Step 5 Click on the **Phone Number** drop-down menu and select the external phone number where calls will be routed. (This can be your cell, home or other PSTN number that is not within the ShoreTel 6.1 system.) If the external number has not already been entered, you can do so now.

You cannot configure a PSTN number that would require the call to go out a SIP trunk.

Step 6 Select the desired **Activation Settings** radio button. Office Anywhere initiates a dialog at the external number prompting the user to press their keypad "1" to accept the call.

- **Wait for Answer** - The calling party will be connected to the OA user as soon as he answers his cell or home phone.
- **Wait for DTMF** - The calling party will be connected to the OA after the OA user touches the appropriate DTMF key.

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

Step 8 Click the **Options** menu and select **Configure Call Handling**.

Step 9 In the **Call Forwarding Condition** area of the window, select the **When No Answer or Busy** radio button (if it has not already been selected).

Step 10 Under the **Call Forwarding Destination** area of the window, click the **Edit Destination** button to display a window similar to below:

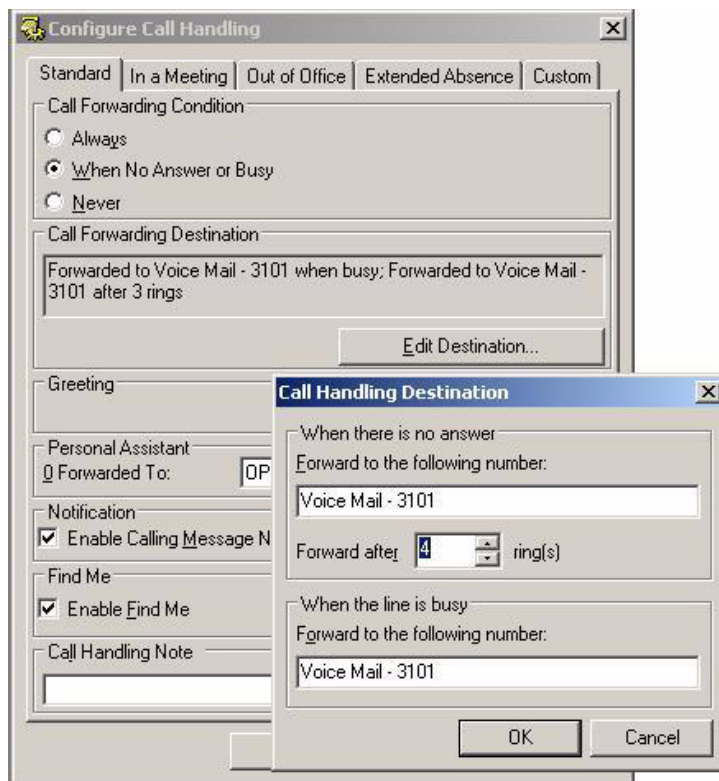


Figure A-4 No Answer Number of Rings

Step 11 In the **Forward after x rings**, select 4 or more rings. (The default, 3 rings, may make it difficult to answer calls before they are sent to voicemail.)

Step 12 Click OK.

Details:

- When finished, you can verify that Office Anywhere is enabled on your client machine by looking for the cell phone icon at the bottom right corner of the PCM window (see below).
- By hovering over the cell phone icon, a pop-up window displays the phone number you entered above, indicating that the phone number has been successfully mapped to your this extension.

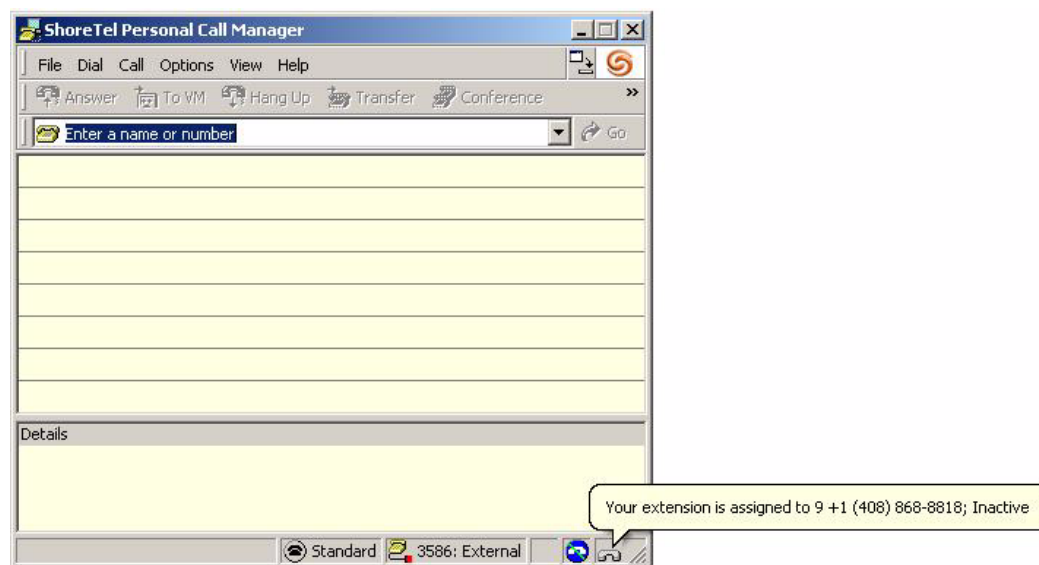


Figure A-5 Verifying proper PCM configuration

A.5 Configuring On-Net Extension Assignment

On-net extension assignment is intended for users who may travel frequently, accessing the ShoreTel 6.1 system from multiple sites “on the network.” These users can benefit from not being tied down to a single physical telephone, and Office Anywhere allows them the freedom to pick up a ShoreTel telephone and assign their extension to that telephone via the voicemail menu.

To assign or unassign an extension to any on-net telephone using the voicemail Telephone User Interface (below) or using PCM (see Section A.5.2), follow the appropriate procedure below:

A.5.1 Using the Telephone User Interface

To assign an extension to a telephone:

Step 1 Log in to voice mail.

Step 2 Press 7 to select *Change Mailbox Options*.

Step 3 Press **3** to select *Re-assign Extension*.

Step 4 Press **1** to select *Assign*.

Step 5 Wait for a dial tone, then hang up.

This option is available only from telephone ports and not trunk ports.

To unassign an extension from a telephone:

Step 1 Log in to voice mail.

Step 2 Press **7** to select *Change Mailbox Options*.

Step 3 Press **3** to select *Re-assign Extension*.

Step 4 Press **2** to select *Unassign*.

Step 5 Wait for a dial tone, then hang up.

If no other user is assigned to the home telephone port, the extension automatically reverts back to the home telephone. If another user is assigned to the home telephone port, the extension will be assigned to the SoftSwitch until the home telephone port becomes available. A user can “kick out” the other user from the home telephone port by assigning the extension from their home telephone using the procedure above.

A.5.2 Using the ShoreTel Call Manager

The ShoreTel Call Manager login is integrated into Microsoft Windows. If you want to run the ShoreTel Call Manager from another workstation, you must exit Windows, and log in using your profile. You must have a valid profile on the workstation, as well as a properly installed ShoreTel Call Manager.

To return to a home telephone port:

Step 1 Verify the user status displayed in the lower-right corner of PCM.

Step 2 Click the **Go Home** icon on the status bar, as shown below, or select **Go Home** from the **File** menu.



Figure A-6 Office Anywhere Status and Go Home Function from the Status Bar of PCM

Emergency 911 Operations

The following sections are included in the chapter:

- Overview
- Feature Operation
- Selecting Caller ID Type for 911 Calls
- Configuring Your System
- International Emergency Numbers
- Verifying Your 911 Configuration
- Additional Recommendations

B.1 Overview

This chapter explains the chain of events in the call flow when a 911 call is placed. This chapter also provides instructions for configuring your ShoreTel 6.1 system to ensure that emergency services are dispatched to the correct location. And finally, the chapter tells you how to select which of the various pieces of caller ID information will be used to identify callers when a 911 call is placed.

B.1.1 How 911 Calls Work

Figure B-1 displays a simple 911 call-flow scenario.

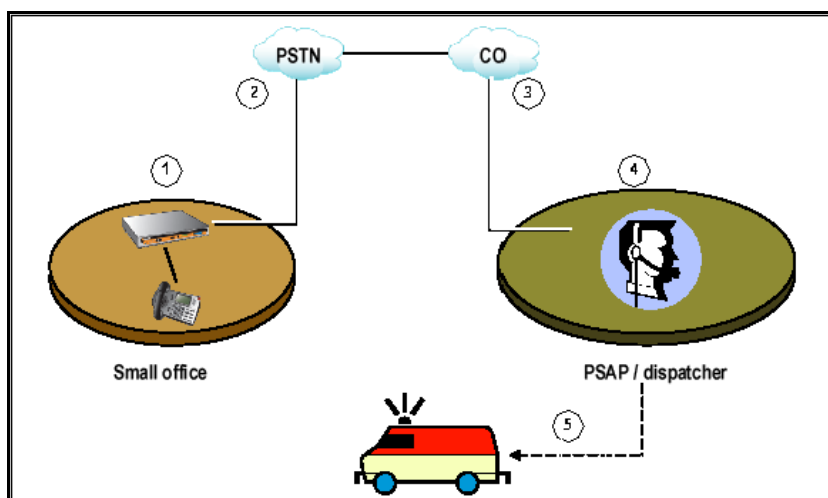


Figure B-1 Simplified 911 Call Flow Scenario

The following is a description of the call flow depicted in Figure B-1.

1. A 911 call is placed from an office and is routed through the ShoreTel 6.1 equipment to an outbound trunk. Caller ID information is provided by the PBX (for PRI trunks) or by the service provider (for non-PRI trunks).
2. The Public Switched Telephone Network (PSTN) forwards the call to the service provider's Central Office (CO).
3. The CO then passes the call to a Public Safety Answering Point (PSAP). This is the location of the emergency services dispatcher.
4. The dispatcher at the PSAP gets a "screen pop" which displays information contained in a 911 database. The database contains a mapping between the caller ID number and the geographic location of the caller.
5. The dispatcher sends emergency response personnel to the calling party's location.

For 911 calls placed from residential or a single-site businesses, determining the location of the calling party is fairly simple and straightforward. However, when dealing with large offices and campus environments, your 911 configuration can get complex. If you are maintaining a configuration that has many remote sites, it is imperative that you do the following:

- keep your 911 information current with your PSAP
- work with your service provider to find out what kinds of caller ID information they will accept
- work with the local PSAP to ensure that any changes in your 911 configuration (i.e. names, phone numbers, locations of the members) are mirrored in the PSAP's database

B.1.2 Roles and Responsibilities

Each participant in a 911 call has a different role to fill and a different set of responsibilities to handle.

The role of the PBX is to:

- route the call out the proper trunk
- pass the correct caller ID number to the CO (PRI only)

The role of the CO is to:

- work with the customer to ensure the correct caller ID number is passed to the PSAP
- pass the "billing number" of the trunk to the PSAP if no other caller ID number is specified

The role of the PSAP is to:

- host a database that maps the caller ID numbers to the physical location of the users
- receive 911 calls
- display information about the calling party to a dispatcher
- send the proper emergency response personnel to the caller's location

The role of the customer (i.e. you) is to:

- decide which type of caller ID information best fits your needs for 911 calls
- work with the service provider to verify that they will accept your preferred type of caller ID information
- communicate any changes to your 911 configuration to ensure the PSAP is current

B.1.3 Selecting a PS/ALI Service Provider

In addition to working with your local PSAP to provide accurate logistical information, we also recommend subscribing to a PS/ALI (Private Switch/Automatic Location Information) service provider.

A PS/ALI service provider maintains a database that stores specific address information for each extension or DID on your system. Subscribing to PS/ALI services ensures that accurate ANI (automatic number identification) information is passed to the PSAP in the event of an emergency call, and prevents the ambulance from showing up at the wrong building!

A subscription to a PS/ALI service provider is particularly recommended in situations where a ShoreTel 6.1 system is deployed in an environment where a single PRI is used to serve multiple locations (for example, if a single PRI is used for several schools in the same district). In such environments, it is possible for a user to make an emergency call from one of the elementary schools and have the emergency crews dispatched to the wrong location. This can happen if the local trunks are busy and the call gets routed across an analog trunk and across the WAN to the first available PRI, which might be at one of the other schools in the district. With no PS/ALI database to provide accurate information about the origination of the call, the emergency services providers see the call originating at the wrong location. While the correct phone number is sent to emergency services, the association is with the PRI instead of the school where the call originated.

This critical error can be prevented if a PS/ALI database is in place. Such a database (maintained by a PS/ALI service provider) can identify the location associated with a specific DID.

Note that ShoreTel 6.1 does not provide PS/ALI service. Please contact your local Telco carrier, who should be able to provide a list of PS/ALI service providers (e.g. RedSky, 911 Enable and Teltronics, etc.).

B.2 Feature Operation

B.2.1 Digit Collection for 911 Calls

A ShoreTel user who dials 911 or <access_code> + 911 will be routed to a 911-capable trunk.

- If the user dials an access code followed by 911, digit collection terminates immediately and the call is routed to a 911-capable trunk.
- If the user forgets to dial an access code before dialing 911, the system waits five seconds before routing the call to a 911-capable trunk. This pause has been introduced to eliminate accidental calls to 911. Most systems use an access code of 9 for outbound calls, making it easy for a user to mistakenly dial 911 on a long-distance call by adding an extra 1 before the area code (for example, if he or she dialed the following number 9-1-1-408-555-1212). If additional digits are entered after 9-1-1 during the five-second timeout period, the system will consider it a dialing error and the calling party will hear a reorder tone.

B.2.2 Ensuring Proper Routing of 911 Calls

Call permissions are ignored when a 911 call is placed to ensure that a user can dial 911 from any extension on the system, regardless of the permissions associated with that user or the extension from which he or she is calling.

Once the user dials 911, the call leaves the extension, arrives at the switch, and is routed to any available 911-capable trunk at the originating site. If the user belongs to a user group that does not have access to any 911-capable trunks, then the call will not be placed.

WARNING When adding users to the ShoreTel 6.1 system, make sure each user is placed in a user group that has access to a 911-capable trunk group. If a user is placed in a user group that does not have access to a 911-capable trunk (e.g. a user group with long-distance trunks only), members of that user group will not be able to dial 911, and they will get reorder tone when attempting to do so.

To better understand this, you must realize that users are placed into user groups when added to the ShoreTel 6.1 system. The user groups are assigned to trunk groups, and these trunk groups have different capabilities, one of which is the ability to place 911 calls. If a user belongs to only one user group, that group must have access to a 911-capable trunk. It is crucial that each site have at least one 911-capable trunk.

For details on adding users to a user group that has access to a 911-capable trunk, see Section 10.3 on page 185.

Always confirm with your service provider that a trunk supports 911 calls. In some instances, this may not be the case (for example, with long-distance trunks). If the trunk does not support 911, be sure to un-check 911 as an available service in the associated trunk group in Director.

If you have mistakenly set up a site that has no available 911-capable trunks, 911 calls will be routed to the 911-capable trunk at the proxy site if one has been designated. By routing the call to a proxy site, the ShoreTel 6.1 system is making a “last ditch” attempt to place the 911 call. This failover behavior can be unreliable and should not be relied upon to ensure that users on your system can dial 911. If you use the “parent as proxy” configuration, make sure the boundary between the two sites never traverses geographic locations that would send a 911 call to the incorrect 911 service provider. For example, if improperly configured, a caller in Houston could pick up a phone, dial 911, and reach a 911 service in Boston because the system was configured to have the Boston site as the parent of the Houston site with “parent as proxy” checked.

Each site should have at least one 911-capable trunk. If there will only be one trunk at a particular site, that trunk should be capable of placing a 911 call. You should also be aware that if there is only one trunk at a site, only one 911 call can be placed at a time. Therefore, you should make sure you have enough 911 trunks at each site to accommodate the (realistic) potential 911 traffic for that site.

B.2.3 Trunk Signaling for 911 Calls

When a 911 call is routed out an analog or digital loop-start or a digital wink start trunk, the service provider is responsible for passing caller ID information to the PSAP.

When a 911 call is routed through a T1 PRI trunk, the ShoreTel 6.1 System sends the proper caller ID information to the service provider, and the service provider must forward the information to the PSAP.

Contact your local telecommunications service provider to communicate your 911 implementation plans and have them approved. It is important to ensure that the service provider will accept, and subsequently pass to the PSAP, the caller ID information

configured within the ShoreTel 6.1 system. In some cases, without proper planning, a provider will reject the caller ID information as configured in the ShoreTel 6.1 system and will simply pass the caller ID information associated with the trunk to the PSAP. If this happens, the dispatcher may get a number telling them to go to the wrong location.

User's have a "home port" defined in ShoreWare Director. If a user is not at his home port, it could change the caller ID number delivered to the service provider on 911 calls.

For mobile workers who travel between sites, the user must have access to a 911-capable trunk at every site. In remote locations, the user should use the 911 trunk associated with that remote location.

B.3 Selecting Caller ID Type for 911 Calls

There are a number of different caller ID choices available within ShoreWare that can be used by the PSAP to identify callers when they place a 911 call. The list below summarizes the available choices for sending the caller ID to the service provider for 911 calls. Options are listed in the order of precedence, meaning that if the first item on this list is not configured within the ShoreTel 6.1 system, then the next piece of information on the list will be sent. Additional details about each of these caller ID options appears after the list.

1. User's Caller ID number
2. User's Direct Inward Dialing (DID) number
3. Caller's Emergency Service Identification ID (CESID) for an IP address range
4. CESID of the controlling switch
5. CESID of the site
6. Nothing sent by ShoreTel 6.1 system (the service provider sends the caller ID number associated with the trunk)

For details on selecting the best choice for your situation, refer to Section B.3.1.

If you are configuring a system in the Netherlands, please see Section B.6.1 on page 412.

B.3.1 Available Caller ID Options

User's Caller ID number – Each user can be assigned a caller ID number that will identify him during outbound calls. This caller ID number is typically used for outbound calls from the ShoreTel 6.1 system when you do not want the receiving party to know the calling party's DID number. For example, an ACD agent may use caller ID to ensure that returned calls will go to a queue of sales agents, rather than directly to his desk. Similarly, this caller ID number can be sent to the service provider to identify the user when he places an outbound 911 call. The user's caller ID number is a very specific way of identifying the location of an individual user and is therefore likely to become less accurate over time as the PSAP's 911 database becomes out of date. Sending the CESID for outbound 911 calls is best for smaller organizations (see Figure B-2) and is defined on the User Edit page. You must select the "Send Caller ID as Caller's Emergency Service Identification" check box on the user group page.

In the scenarios described above and below, the user's caller ID number will only be sent when the user is at his home port. If the user is not at his home port, then the next available caller ID type is sent.

User's DID number – The DID number (Direct Inward Dialing) is the number someone dials from outside the ShoreTel 6.1 system to reach a user at her desk. The DID is what most people would consider to be a “normal” telephone number. This DID number can be sent to the service provider to identify the user when she places an outbound 911 call. Although this is the most granular way of identifying users, it is also the most likely to become out of date in the PSAP's 911 database as people come and go. Sending the DID number for outbound 911 calls is most appropriate for smaller organizations (see Figure B-2) and is defined on the User Edit page. You must select the “Send DID as Caller's Emergency Service Identification” check box on the user group page.

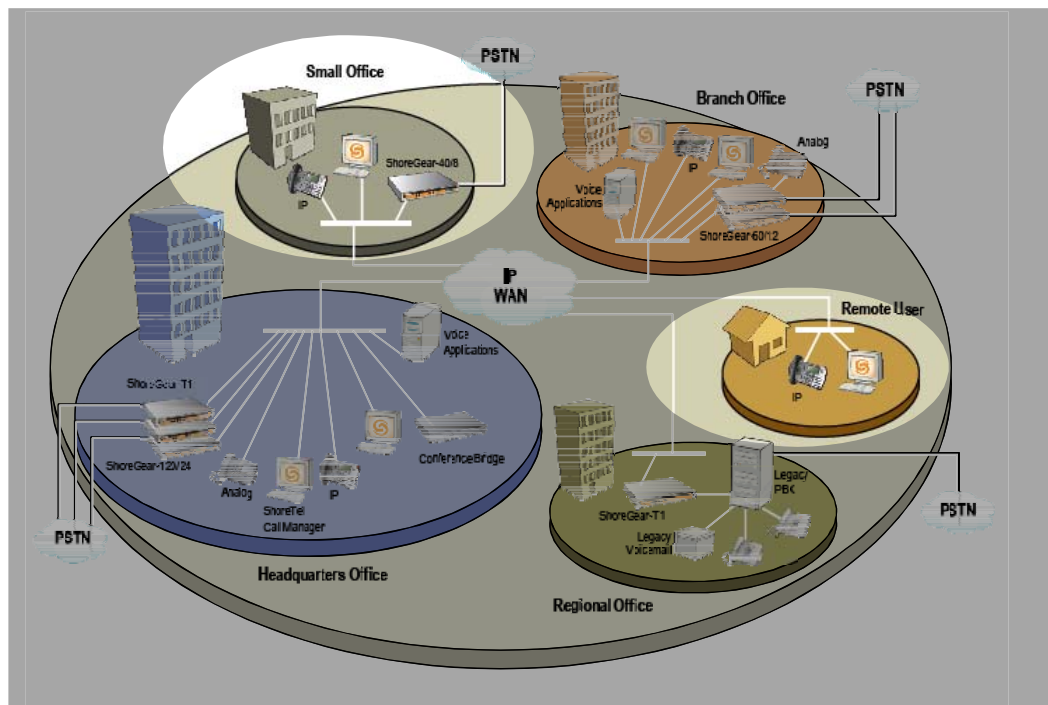


Figure B-2 Caller ID Options for Small Offices

CESID of the Specified IP Address Range – The CESID of an IP address range can also be delivered to the service provider during outbound 911 calls. A single CESID number is assigned to a range of IP addresses such that any IP phone that has an IP address that falls within the specified range will have this CESID sent for outbound 911 calls. This option works best for identifying a phone in an office that has many floors and many extensions. Typically, a specific IP address range is configured for each floor of a building so that all users on that floor use the same CESID for 911 calls.

If a DHCP server is present, an IP phone will automatically receive an IP address within the specified range when it is connected to the network.

Sending the CESID for a specified IP address range for outbound 911 calls works best for larger organizations where simply identifying the site's street address would not provide enough information for an emergency response team to locate the caller (see Figure B-3). Furthermore, this option offers the best flexibility, the highest accuracy, and is the least likely to become out of date in the PSAP's 911 database. This option is defined on the IP Phone Address Map page.

CESID of the controlling switch – Similar to the previous option, the Caller's Emergency Service Identification ID (CESID) of the controlling switch can also be sent to the service provider during outbound 911 calls. With this option, a CESID number is assigned to a

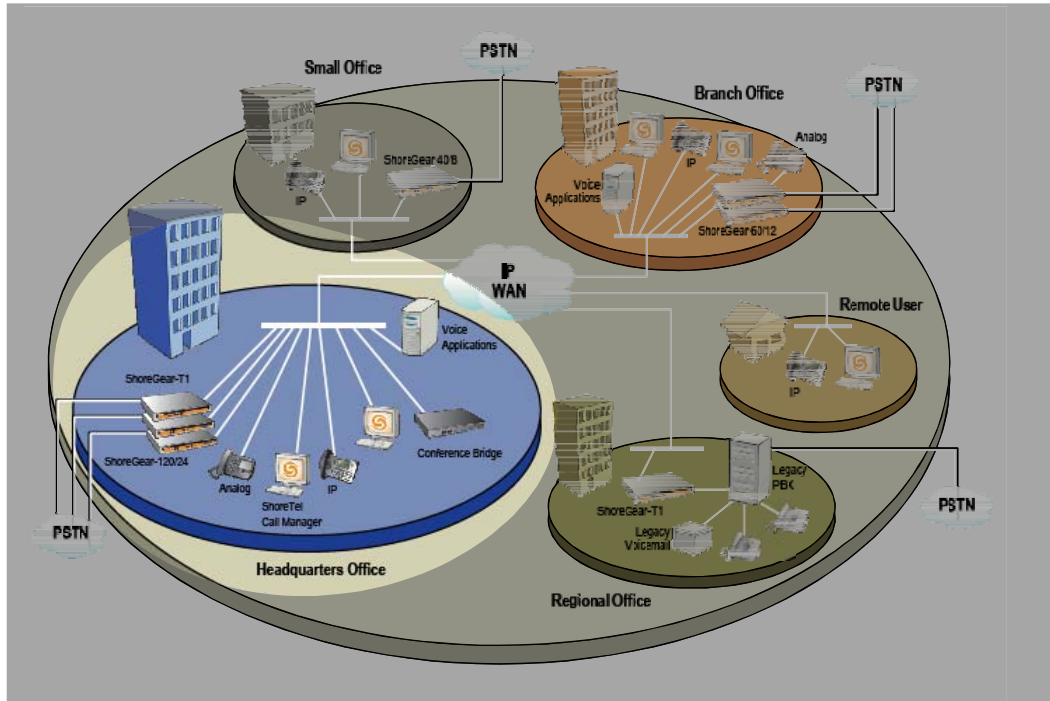


Figure B-3 IP Address Mapping – Best for Larger Offices

phone switch and for any phone plugged into this switch, the switch's CESID is sent for outbound 911 calls. This option is best for larger organizations in which users are calling from analog phones (see Figure B-3). Using the IP Phone Address Map method will not work with analog phones. This approach ensures that the emergency response team is sent to the approximate vicinity of the calling party. This option is defined on the Switch Edit page.

Site (Caller's Emergency Service Identification (CESID)) – This option delivers the CESID associated with the site to the service provider during 911 calls. This approach might not be granular enough for larger enterprises, but it could work well for single-site organizations or for situations in which it would be adequate to provide the emergency response personnel with a building address. This option is defined on the Site Edit page. (See Figure B-4).

Table B-1 shows several common customer scenarios and provides recommendations for how E911 should be configured, along with reasons why that is the preferred configuration.

Rules and regulations for E911 vary from one region to the next. Check with the local public safety agency to ensure you configure the system to meet the necessary requirements.

B.4 Configuring Your System

The following subsections provide information for configuring your ShoreTel 6.1 system for proper Emergency 911 operation.

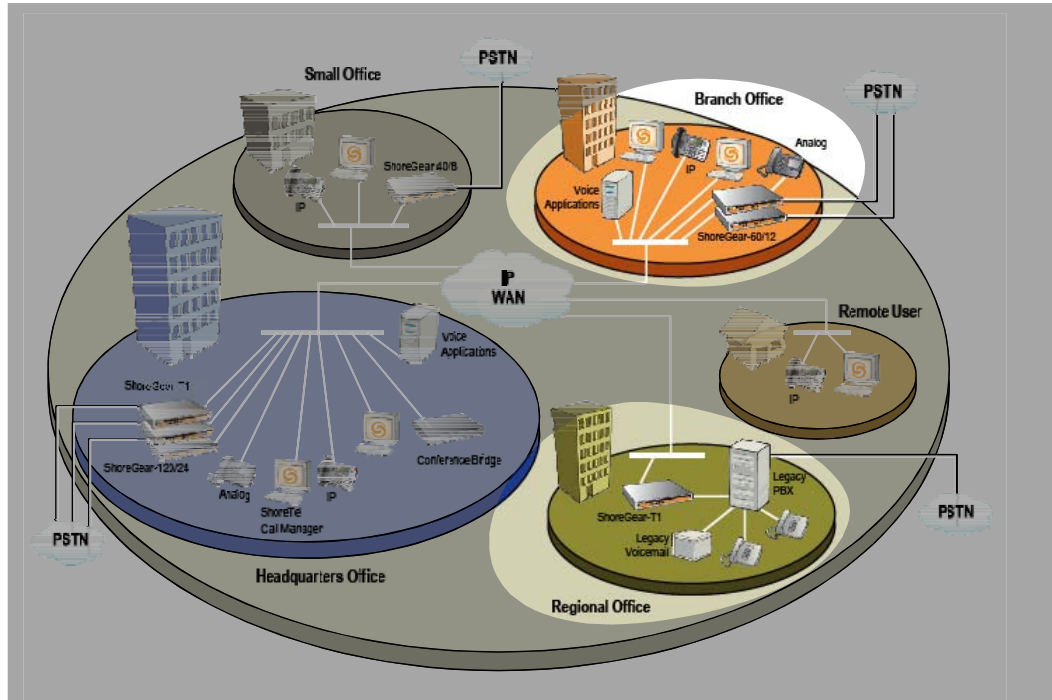


Figure B-4 Send the Site CESID for Medium-Sized Offices

Scenario	Note
Small site with analog trunks	<ul style="list-style-type: none"> No 911 configuration necessary
College dormitory rooms (with PRI)	<ul style="list-style-type: none"> Emergency response personnel must be dispatched to a specific room Consider sending Caller ID or DID (we recommend turning off extension assignment for this application)
Classroom (with PRI)	<ul style="list-style-type: none"> Emergency response personnel must be dispatched to a specific room Send DID or Caller ID (Consider turning off extension assignment)
Multi-building campus or office complex (with centralized PRI)	<ul style="list-style-type: none"> Caller ID or DID may be too granular with too much management overhead Consider using IP phone address mapping and/or the switch CESID
Large building with multiple floors (with PRI)	<ul style="list-style-type: none"> Caller ID or DID may be too complex Consider using IP phone address mapping and/or the switch CESID
SoftPhones or travelling user	<ul style="list-style-type: none"> Use home phone or hotel phone for 911 calls
Remote IP phones (with PRI at headquarters)	<ul style="list-style-type: none"> Dial 911 with home phone Use the IP phone address map (home CESID) as a backup
VPN Phone – Fixed Location	<ul style="list-style-type: none"> Remote worker install phone once, then never moves it Configure Caller ID of phone to reflect geographic location. One option is setting Caller ID to be identical to worker’s home phone number
VPN Phone – Variable Location	<ul style="list-style-type: none"> Remote worker uses phone when traveling from various locations Use home phone or hotel phone for 911 calls.

Table B-1 E911 configuration options

B.4.1 Trunk Groups

Make sure you have an outbound trunk group with outbound access that also supports the 911 trunk service. If there is no 911-capable trunk group configured, you should create one on the appropriate *Trunk Group* edit page (see Figure B-5). See Chapter 7, starting on page 95, for trunk group configuration information.

The screenshot shows the 'Trunk Groups' configuration page for a 'T1 PRI Trunk Group'. The page includes several sections:

- General Information:** Name (PRI), Site (Headquarters), Language (English).
- Inbound:** Number of Digits from CC (10), DNIS, DID, Extension, Tandem Trunking, User Group (Executive), Dial In Prefix, Destination (4700 - Default).
- Outbound:** Network Call Routing, Access Code (9), Local Area Code (408), Additional Local Area Codes, Nearby Area Codes.
- Trunk Services:** Local, Long Distance, International, n11 (e.g. 411, 611), **911** (highlighted), Easy Recognizable Codes (ERC), Explicit Carrier Selection.

Figure B-5 Trunk Group Edit Page (T1 PRI Trunk Group)

As a precaution, you should review all other trunk groups to make sure that the 911 check box is not inadvertently enabled on a trunk that is not 911-capable.

B.4.2 User Groups

Make sure each user group has access to a 911-capable trunk group. You can select the desired 911 Caller ID choice on the *User Group* edit page.

- To send the Caller ID as the CESID number, verify the **Send Caller ID as Caller's Emergency** check box is selected.
- To send the DID as the CESID number, verify the **Send DID as Caller's Emergency** check box is selected.

See Section 10.3 on page 185 for more information about configuring user groups.

Make sure you give access to trunk groups at other sites in case users in the group use the Office Anywhere feature from another site.

Figure B-6 User Group Edit Page

B.4.3 Users

Make sure the Caller ID field is configured if you are sending Caller ID as CESID for this user. Similarly, make sure the DID check box is selected (and contains a valid number in the DID field) if you are sending DID as CESID for this user.

Verify each user belongs to the correct user group. Use the *Edit User* page (Figure B-7) to associate users with the appropriate user group. See Section 10.4 on page 187 for more configuration information.

Figure B-7 Edit User Page

You cannot configure any user (including workgroups or route points) with a 911, 911x, or 911xx extension. These extension ranges are reserved for the 911 feature operation.

B.4.4 IP Phone Address Map

Use the IP Phone Address Map (Figure B-8) to assign a CESID to a range of IP addresses. See Section 3.3 on page 40 for more configuration information.

The screenshot shows a web browser window with the URL `http://10.2.28.12 - IP Phone Address Map Info - Microsoft Internet Explorer`. The form contains the following fields and values:

- Site: Remote
- Low IP Address: 10.2.28.18
- High IP Address: 10.2.28.18
- Caller's Emergency Service Identification (CESID): 14083313030 (e.g. +1 (408) 331-3300)
- Teleworkers:

Buttons at the bottom include Save, Close, Previous, and Next. A copyright notice at the bottom reads: © 1998-2005 ShoreTel, Inc. All rights reserved.

Figure B-8 IP Phone Address Map

B.4.5 Switch

Use the Switch edit page (Figure B-9) to configure a CESID number for a switch for outbound 911 calls. See Section 5.4.2 on page 60 for more configuration information.

The screenshot shows a web browser window with the URL `Switches Edit ShoreGear-120/24 Switch`. The form contains the following fields and values:

- Name: SG-24
- Description: Sunnyvale
- Site: Headquarters
- IP Address: 10.2.28.20
- Ethernet Address: 00-10-49-00-0F-32
- Save to Manage Switch: Headquarters
- Caller's Emergency Service Identification (CESID): 14083313020 (e.g. +1 (408) 331-3300)
- Music On Hold Source:

Buttons at the top include New, Copy, Save, Delete, and Reset. A 'Find Switches' button is located next to the IP Address field.

Figure B-9 Switch Edit Page

B.4.6 Sites

Use the Site edit page (Figure B-10) to configure a site's CESID number. See Chapter 3, starting on page 39, for additional configuration information.

In the United States and Canada, make sure the **Emergency Number** field is set to 911. The emergency number is configurable by site since it can vary by country.

The screenshot shows the 'Site Edit Page' with the following fields and values:

- Name: Headquarters
- Country: United States of America
- Language: English(US)
- Parent: Top of Tree
- Use Parent As Proxy:
- Extension Prefix: 888
- Local Area Code: 488
- Additional Local Area Codes: Edit
- Caller's Emergency Service Identification (CES ID): +1 (702) 385-4997 (e.g. +1 (408) 321-330)
- Time Zone: (GMT-08:00) Pacific Time (US & Canada, Pacific Standard Time)
- Night Bell Extension: 888-5500
- Night Bell Switch: 124-Fuji24-hq Edit Night Bell Call Handling
- Paging Extension: 000-55001
- Paging Switch: 124-Fuji24-hq
- Operator Extension: Search
- FAX Redirect Extension: Search
- Admission Control Bandwidth: 1544 kbps
- Emergency Number List:
 - Trunk Access Code Required 911
 - Trunk Access Code Required 112
 - Trunk Access Code Required 533 Add More...

Figure B-10 Site Edit Page

B.5 Planning Your Emergency Response

When a 911 call is made, the system automatically generates an event in the Windows event log at the beginning of the call. With the use of an event filter, you can automatically send an e-mail message to the appropriate people in your organization to help coordinate your local response (i.e. at the organizational level) whenever 911 is dialed.

We recommend training the personnel at all sites on the emergency operations of your ShoreTel 6.1 IP voice system. All users should know how to access 911 emergency services during normal and power outage situations.

B.5.1 Call Notification

You can set up an event filter to generate an e-mail message to help coordinate your emergency response. Use the *Event Filter* edit page (Figure B-11) to configure the event filter for the following parameters:

- **Source:** ShoreWare
- **Category:** Switch
- **Event ID:** 1319
- **Type:** All
- **Target Email:** Email addresses or an email distribution list

See Section 18.8 on page 365 for more configuration information.

Figure B-12 shows a typical logging message that would result after a 911 call was placed – assuming notifications had been properly configured.

We suggest naming your switches with location information such that you can understand which site the call was made from.

Figure B-11 Event Filter Edit Page

Figure B-12 911 Event Log

B.6 International Emergency Numbers

The ShoreTel 6.1 system allows dialing of emergency numbers with and without trunk access codes. For this reason, you should reserve the dialing plan space for this feature. For example:

- “112” is used in Europe and other countries.
- “000” is used in Australia.
- “999” is used in Asia.

Thus, to make use of this feature, extensions should not begin with “112”, “911,” or “999”. Note that extensions should never begin with “0”.

Each site can have a maximum of ten emergency numbers to accommodate locations where multiple emergency service numbers are required.

For more information on international installations, see the *ShoreTel 8 Planning and Installation Guide*.

B.6.1 Special Considerations for Netherlands

It is against the law in the Netherlands to "spoof" Caller ID. Caller ID will only be sent if the configured caller ID corresponds to the incoming DID that is associated with a particular trunk.

Thus, any number entered in the CESID field in the "Switch Edit Page" or "Site Edit Page" (see Figure B-9 or Figure B-10), will only be sent if this number matches the number associated with the incoming DID for that trunk.

B.7 Verifying Your 911 Configuration

After you have finished configuring your system for 911 operation, we recommend working with your local 911 dispatch center to test your configuration in order to verify that it has been correctly configured, is sending out the desired caller ID information, and is dispatching emergency response personnel to the proper location.

We recommend calling your local law enforcement agency's non-emergency number to understand how to go about the test and to arrange a call time during non-peak hours. Do not place your 911 test call without making prior arrangements! Depending on your location, an officer may be required on-site when making test calls.

Table B-2 is intended to help you plan your test call to the local dispatch center.

Site	Extension	User	Expected Caller ID	Actual Caller ID	Pass or Fail

Table B-2 Emergency Call Test Matrix

B.8 Additional Recommendations

All sites should be configured with a designated power failure emergency phone configured appropriately. Each designated power failure emergency phone should be configured on the following ports, based on type of switch, to take advantage of ShoreTel 6.1's emergency line power failure feature:

- SG40 – Port 4: Analog Trunk; Port 5: Analog Emergency Phone
- SG60 – Port 8: Analog Trunk; Port 9: Analog Emergency Phone
- SG120 – Port 8: Analog Trunk; Port 9: Analog Emergency Phone
- SG30 – Port 1: Analog Trunk; Port 12: Analog Emergency Phone
- SG50 – Port 1: Analog Trunk; Port 12: Analog Emergency Phone
- SG50V – Port 1: Analog Trunk; Port 12: Analog Emergency Phone
- SG90 – Port 1: Analog Trunk; Port 12: Analog Emergency Phone
- SG90V – Port 1: Analog Trunk; Port 12: Analog Emergency Phone

- SG220T1A – Port 1: Analog Trunk; Port 12: Analog Emergency Phone

Call Detail Record Reports

Call detail record (CDR) reports are used to review the ongoing call activity on the ShoreTel 6.1 system. Sections in this appendix include:

- Overview
- MySQL Database
- CDR database
- Generating a CDR Report
- Interpreting CDR Reports

C.1 Overview

The ShoreTel 6.1 system tracks all call activity and generates call detail records into a database as well as into a text file on the ShoreWare server. The call detail records are used to generate CDR reports. The system comes bundled with twelve CDR reports based on data from the CDR database. In addition, the text files provide third-party call accounting systems with a simple and standard way to access the call data.

If the ShoreWare server is not up and running, call detail records will not be generated, and the associated calls cannot be presented in the CDR reports.

In the event of a WAN outage, CDR data is stored for up to two hours on the distributed server. When WAN connectivity is restored, the stored data is forwarded to the Headquarters database. After two hours, the distributed server deletes the data and logs an error to the NT event log.

C.1.1 Call Accounting Service

The ShoreTel 6.1 system has a call accounting service running on the main server that generates call detail records into a database, as well as into a space-delimited text file for use by third-party call accounting applications. The call accounting service is also responsible for archiving all the CDR data.

The CDR files are located on the ShoreWare server in the following folders:

- ShoreTel Data / Call Records 1—This contains call records for releases prior to Shoreline3 Release 3.1k.
- ShoreTel Data / Call Records 2—This contains call records for Shoreline3 Release 3.1k and later releases.

C.1.2 Active CDR Database

The call accounting service generates call detail records into the active CDR database. This file includes all call activity for the period of time specified in the *Retention Period for CDR Data* parameter in the Director Reporting Options panel, as shown in Figure C-1. To access this panel, select *Reporting -> Options* from the Director menu.



Figure C-1 Reporting Options panel

When *Enable Archiving* is selected on the Reporting Options panel, a nightly routine automatically moves call detail records that are older than the limit specified by the *Retention Period for CDR Data* into the Archive database.

Refer to Section C.2.2 on page 418 for instructions on setting up the Archive database.

C.1.3 Legacy CDR Text Files

The call accounting service automatically generates a daily legacy CDR text file for use by third-party call accounting applications. These packages typically provide numerous reports, including:

- Call accounting, cost allocation
- Most frequently dialed numbers
- Most costly dialed numbers
- Most costly users
- Trunk utilization
- Toll fraud

The CDR*.log files are text files created daily at midnight. It contains call records from midnight to midnight. Any call records that span the midnight hour will be recorded on the day that calls are completed.

C.1.3.1 Format

The file name format for the daily legacy CDR log file is “CDR-MMDDYY.log” where “MM,” “DD,” and “YY” are two-character, zero-padded month, day, and year numbers respectively.

Call records are entered in the log file in the order of when the call was completed and not when it began.

It is the responsibility of the third-party reporting application to delete the daily log files.

The format of the record is column based, must be justified correctly, and end with a carriage return and line feed. A single blank character is inserted between each data field for readability.

The following are the field definitions:

Field	Column	Length	Comment
Call ID	1	10 right justified	A unique ID that represents the call. The Call ID is meant to be unique for the duration while it's active.
Date	12	10	Date of the call given in month, day, and year: mm/dd/yyyy
Time	23	8	Start of the call given in hours, minutes, and seconds: hh:mm:ss
Extension	32	16 left justified	Inbound or outbound extension ID. Last valid party on the call. Valid parties include user extension and operator but not voice mail or auto-attendant.
Duration	49	8	Call duration given in hours, minutes, and seconds. hh:mm:ss
Call Direction	58	1	Incoming/outgoing flag 0 – Incoming 1 – Outgoing 2 – Tandem Trunking – Inbound Tandem Call 3 – Tandem Trunking – Outbound Tandem Call
Dialed Number	60	16 left justified	Contains the number dialed but does not include any access code, such as 9, to seize the trunk. Valid only for outbound calls.
Caller ID	77	16 left justified	Blocked or unavailable information will be reported as blocked or unavailable in text. Valid only for incoming calls.
Trunk Member	94	4 right justified	The Port ID of the trunk.
Trunk Group	99	3 right justified	The Trunk group ID.
Account Code	103	20	The account code entered by the caller.
CR /LF	124		Carriage return.

Table C-1 CDR Text File Field Definitions

C.2 MySQL Database

CDR records are maintained and queried through MySQL database. The maximum MySQL database size is 64 terabytes (TB) with database tables restricted to a maximum size of 2 TB.

The data in the MySQL files can be viewed using a new Web-Based Reporting feature from ShoreWare Director (see Section 20.2 on page 384 for more information). Alternatively, the user can use common database command utilities via a command line interface to dump and restore files.

MySQL was introduced to ShoreWare in version 7.0. Previous versions of ShoreWare stored CDR records using Microsoft Joint Engine Technology (JET) database management. After ShoreWare version 7.0 is installed, call detail records are stored in the MySQL database. Systems upgraded from a prior release maintains the previous database stored in the Microsoft format.

When a CDR report is generated on an upgraded system, only the MySQL database will be tapped to generate the report. Data in the older Microsoft database is ignored until it has been converted to the MySQL format. This process is described in Section C.2.4 on page 422.

CDR reports are generated from Director, as described in Section 20.2 on page 384.

MySQL database service can be monitored or restarted from Director by selecting *Maintenance -> Services* in the menu panel, then selecting MySQL in the table on the Services page.

C.2.1 Compatibility and Pre-Configuration Requirements

C.2.1.1 Disk Space Requirements

Storing call detail records for 50,000 workgroup calls requires a 1.5 GB MySQL database. Implementing a database of this size typically requires 4.0 GB of disk space. This includes disk space for the main database (1.5 GB), the archive database (1.5 GB), and temporary space required to generate reports (1.0 GB).

Although the main and archive databases are typically stored on the same server, MySQL permits the storage of the databases on different servers.

C.2.1.2 Compatibility with Utility Programs

ShoreWare should be run on a dedicated server. Other programs that access MySQL databases may not be compatible with ShoreWare, resulting in installation and data integrity issues. Before installing ShoreWare on any server, remove all pre-existing MySQL programs and databases.

Virus Checkers: MySQL database files must be excluded from all virus checker utilities running on the server. Specifically, if virus checker is running on the server, exclude MySQL CDR Database file (c:\Shoreline Data\Call Records 2\Data\[ibdata1, ib_logfile0, ib_logfile1]) from your virus checker utility. If these files are not in the exclusion list, MySQL service will stop working.

Disk or Backup utilities: MySQL database files must be excluded from all disk or backup utilities running on the server. Failure to exclude the database will crash the MySQL service.

To restart the database after a crash, access the MySQL Service panel from ShoreWare Director by selecting *Maintenance -> Services* in the menu panel, then selecting MySQL in the table on the Services page.

C.2.2 Archival and Backup Utilities

The following section describes database archival, backup, and replication tools. Table C-2 summarizes the service availability for these features.

Field	Backup	Archive (Secondary Server)	Replication
Technical Assistance Support	Yes	Yes	No
Additional MySQL License Required	No	Yes	Yes

Table C-2 Archival, Backup, and Replication Services Availability

Field	Backup	Archive (Secondary Server)	Replication
Execution	Manual	Daily	Online
Reports run on remote machine	No	Yes	No
Complete restoration if HQ fails	Yes	possible through manual recovery	Yes

Table C-2 Archival, Backup, and Replication Services Availability

C.2.2.1 Record Retention Periods

You configure the number of days that a database stores a Call Detail Record in the Report Options panel. To access the Report Options panel, as shown in Figure C-2, open ShoreWare Director and select *Reporting -> Options* from the Director Menu.

- *Retention Period for CDR Data* specifies the number of days that records remain in the main CDR database. Older records are removed from the database each day.
- *Retention Period for CDR Archive* specifies the number of days that records remain in the archive database. Older records are removed from the database each day.

Default values for each parameter is 125 days.

Figure C-2 Report Options panel

C.2.2.2 Database Archive Utility

The archive utility provides a method of removing older records from the main database and storing them in an archive database. Archiving older records into a separate database reduces the storage requirements of the main database, which reduces the time required to search for specific records or generate reports. The archived database provides the same set of services as the main database.

Archival services are configured and enabled in ShoreWare director, where you can specify the number of days that records are maintained in the main database and in the archive database. When archiving is enabled, archival services are performed daily. The archival service copies records to the archive database that exceed the main database age limit, then removes those records from the main database. Records that exceed the age limit for the archive database are removed from the archive database. Age limits are established separately for each database; valid limits range from one to 2000 days. The default age limits for each database is 125 days.

Example: A sample implementation sets a 30 day limit on the main database and a 365 day limit on the archive database. In this case, the main database contains records for calls handled during the past 30 days while the archive database contains records for calls handled during the past 365 days.

The Backup utility can be used for record storage requirements that exceed 2000 days.

To create an archive database:

- Step 1** Run `MakeCDRArchive -d databasename`, where `databasename` is the name of the archive database to be created,
- Step 2** Access the Reporting Options panel in ShoreWare Director (Reporting | Options from the Menu panel) to configure ShoreWare to access the archive database.

C.2.2.3 Database Backup Utility

The Backup utility creates a copy of a specified database, which can be restored at a later time and different location. The Backup utility differs from the Archival utility as follows:

- Archiving is configured once then performed daily. Backups are performed only when a command is executed.
- Archival operations are configured from ShoreWare director. Backups are performed from the command line.
- Archive databases can be accessed directly to generate reports. Backup databases must be restored before performing search and report generation tasks.

Backup and Restore operations can be performed without shutting down the MySQL service. Performing these operations during off peak hours reduces the execution time and the impact on other system services.

The file located at "C:\Program Files\Shoreline Communications\Shoreware Server\MySQL\MySQL Server 5.0\Examples\dump1.bat" is an example of a batch file that backs up a MySQL CDR database under generic default conditions. This file can be used as a template for creating a batch file that backs up the database under specific conditions. The password is shorewardba. Backing up a 1.5 GB database requires 200 seconds.

Refer to <http://dev.mysql.com/doc/refman/5.0/en/disaster-prevention.html> for MySQL backup tools, add-ons, and documentation.

C.2.2.4 Database Restore Utility

Restoring a database copies the records in the backup database file to the database specified in the restore command. Records in the backup file that are duplicates of records in the target database are listed in the log file and are not restored.

The file located at "C:\Program Files\Shoreline Communications\Shoreware Server\MySQL\MySQL Server 5.0\Examples\restore1.bat" is an example of a batch file that restores a MySQL CDR database under generic default conditions. This file can be used as a template for creating a batch file that restores the database under specific conditions. The password is shorewardba. Restoring a 1.5 GB database requires 1200 seconds.

C.2.2.5 Database Replication

MySQL provides a Database Replication tool. Access the following web pages for more information:

- http://www.howtoforge.com/mysql_database_replication provides MySQL database replication setup information.
- <http://solutions.mysql.com/solutions/tools?type=25> lists tools and add-ons that assist with database replication.

C.2.3 Installing MySQL on a Secondary Server

Although the archive database is normally stored on the main ShoreWare server, MySQL databases can be installed on a Secondary Server to conserve main server resources. A separate licensed copy of MySQL Enterprise Server 5.0 is required to install the database on a Secondary Server.

To install MySQL on a secondary server, perform the following procedure (Replace c:\Program Files\... in the instructions to the location where MySQL is installed on the server).

- Step 1** Install MySQL Enterprise Server 5.0 on a Secondary server.
- Step 2** Select all the default values during installation except for the following items:
 - use root for UserID
 - use shorewaredba for the password,
 - utf8 for the character set as part of the installation
- Step 3** Backup the file c:\Program Files\MySQL\MySQL Server 5.0\my.ini from the Secondary server to a safe location (i.e. c:\MySQL_backup).
- Step 4** Backup the files c:\Program Files\MySQL\MySQL Server 5.0\Data\ib_logfile*] from the Secondary server to a safe location (i.e. c:\MySQL_backup).
- Step 5** Select Start->Administrative Tools->Services->MySQL
- Step 6** Click “Stop the service” and check that MySQL service status is blank
- Step 7** Compare the file from the Main server directory c:\Program Files\Shoreline Communications\ShoreWare Director\example\archive_MySQL_my.ini with a Secondary server file c:\windows\my.ini. Make sure that all the parameters specified in the archive_MySQL_my.ini are set appropriately in my.ini.
- Step 8** Archive_MySQL_my.ini values are as follows.

```
[mysql]
default-character-set=utf8
[mysqld]
default-character-set=utf8
tmp_table_size = 30M
key_buffer_size=2M
read_buffer_size=2M
read_rnd_buffer_size=2M
sort_buffer_size=2M
innodb_additional_mem_pool_size=2M
innodb_flush_log_at_trx_commit=0
innodb_log_buffer_size=5M
innodb_buffer_pool_size=150M
```

```
innodb_log_file_size=24M
default-storage-engine=INNODB
```

Step 9 Delete the file `ib_logfile*` from the Secondary server directory (`c:\Program Files\MySQL\MySQL Server 5.0\Data`).

Step 10 Check that `innodb_flush_log_at_trx_commit=0` on Secondary server `c:\windows\my.ini` file. If the value is not zero, archiving write operations will be more than 20 times slower.

Step 11 Select Start->Administrative Tools->Services->MySQL

Step 12 Click “Restart the service” and verify that MySQL service comes back up

To convert the Secondary server database into an archive database, perform the following procedure

Step 1 Verify the following files are placed in an equivalent location on the Secondary Server to that on the Main servers (default location is \\Shoreline Communications \ Shoreware Server)

- MakeCDR.dll
- MakeCDR.sql
- MakeCDR_sp.sql
- MackCDRArchive.exe

Step 2 Run `MakeCDRArchive -d databasename`, where `databasename` is the name of the archive database to be created.

Access the Reporting Options panel in ShoreWare Director (Reporting | Options from the Menu panel) to configure the name of the archive database within ShoreWare.

C.2.4 Importing Access Database Files into MySQL Format

When an existing system is upgraded to ShoreTel 7 from a prior release, CDR data is stored in the MySQL format. However, CDR data that was saved prior to the upgrade is stored in the Microsoft Access database files created by the JET database management system. This preexisting data will not appear in the web-based reports generated by Director unless the older data is imported from Microsoft Access to MySQL format.

Importing a 2 GB database (500,000 workgroup calls) requires about 100 minutes on a 2-Quad Core Intel Xeon 2.80 GHz CPU, 4 GB memory, 300 GB HD.

The following procedure describes the process of importing CDR data from a Microsoft Access database into MySQL.

Step 1 1. Set the Retention Period for the database that will receive the records to a value larger than the oldest record that will be imported.

See Section C.2.2.1 on page 419 for instructions on setting the Retention period.

Step 2 Open a command prompt window.

Step 3 Change the command prompt directory to:

```
C:\Program Files\Shoreline Communications \ ShoreWare Server
```


Step 4 Run the following command (parameters are optional):

```
ImportCDR -s server -d databasename -f mdb-file
```

- server is IP address of the mysql shorewarecdr database.
- databasename is shorewarecdr.
- mdb-file is the name of the mdb file (e.g. CDR.mdb).

Details:

- By default, data is imported into the main CDR database on localhost.
- The main CDR.mdb is imported before any of the Archive mdb.
- Errors are written to error log, MakeCDRMMDDYYYY.log. The log will be present in Shoreline Data\Logs.

C.2.4.1 Migrating Custom Reports

Users who wrote custom reports to access MS Access database must modify their custom report generator to work with MySQL CDR database. The user id (st_cdrreport) and password (passwordcdrreport) may be used to read the CDR database.

The CDR reports are located in C:\Program Files\Shoreline Communications\ShoreWare Director\WebReports\Reports. Shoreware reports can be viewed using Crystal Reports XI. Any custom report applications should ensure that they have their process set to either “low” or “below normal” to avoid impacting negatively the call processing performance.

CDR Schema changes in ShoreTel 7

- SIP GUID added to Call table.
- SIP GUID added to MediaStream table.

A_Extension and B_Extension added to Media Stream table – Extension value passed from Tms to TmsCDR. No extension number is available for anonymous calls or for trunk calls.

C.2.5 Performance Tuning for Report Generation

To improve on the CDR report generation performance, increase INNODB_BUFFER_POOL_SIZE defined in c:\windows\my.ini based as specified

Default setting:

- INNODB_BUFFER_POOL_SIZE = 150 MB

If the database contains more than 350,000 records, set

- INNODB_BUFFER_POOL_SIZE = 200 MB

If the database contains more than 500,000 records, set

- INNODB_BUFFER_POOL_SIZE = 250 MB

C.2.6 Report generation time – CPU Utilization

Based on the size of the data requested for the report, the display time for the last page of the report from the first page may require ten minutes. Even though the priority of Report Generation process is set to below normal, generating large reports may potentially impact the call processing performance. To avoid performance degradation issues, do not generate large CDR reports during peak call loads.

C.2.7 MySQL CDR Database and Internationalization

MySQL CDR Database supports the UTF-8 character set. All CDR data in the database is stored in UTF-8 character set.

C.2.8 Monitor MySQL service

To monitor and, when necessary, restart the MySQL service from ShoreWare Director, access the Services page by selecting Maintenance | Services in the menu panel, then select MySQL in the table on the Services page.

C.2.9 Tools for browsing MySQL database tables

MySQL provides MySQL Query Browser as part of their GUI Tools (<http://dev.mysql.com/downloads/gui-tools/5.0.html>). MySQL Query Browser can be used to browse and view the queries. The open source tool (<http://www.webyog.com/en/downloads.php>) is available to view the CDR tables defined in MySQL.

Browsing a large CDR database on the Main server may potentially degrade the call processing server.

Large amount of temporary disk space may be used by these MySQL browser tools. To avoid affecting call processing performance on HQ, a query with LIMIT criteria can be used to show a subset of rows.

C.2.10 Restrictions in the number of records returned by the MySQL CDR query

CDR database queries that exceed 300,000 records may cause performance degradation when generating certain reports, such as Trunk Activity Detail and Trunk Activity Summary. Increasing the amount of free disk space may mitigate this problem, as will modifying the query filter to reduce the number of records returned by the query to under 300,000.

C.2.11 Report License Error Condition

Crystal Reports licensing limits the number of requests that can be simultaneously processed. A request is defined as any user interaction to the report engine that requires report processing, such as first page viewing, printing, drill down, or exporting.

The Crystal Reports license supplied with the ShoreWare Remote Web Reporting License permits a maximum of three simultaneous requests. One license is utilized while the system processes a user's request. When the system has completed processing the request, the license becomes available for another user.

If report requests are terminated before the request completes, the license may not be freed back to the system. There may be instances where a request may result in an "Error occurred reading records: A crystal report job failed because a free license could not be obtained in the time allocated" condition. If this error persists for over 30 minutes, the ReportsAppPool for web reporting in IIS Application pool may require recycling.

IIS ReportsAppPool recycle procedure:

Step 1 Select Start->Administrative Tool->Internet Information Services (IIS) Manager

Step 2 Click Application Pools on the left pane

Step 3 Right click on "ReportsAppPool" and select "Recycle" from the drop down menu.

C.3 CDR Reports

The ShoreTel 6.1 system comes bundled with 12 CDR reports generated using data from the active CDR database on the ShoreWare server. CDR reports present information about users, trunks, WAN links, workgroup queues, account codes, and workgroup agents. Reports are grouped into two categories: summary and detail.

Summary reports provide a high-level view of activity that occurred in a particular area, while detail reports provide a detailed view of activity beyond that of the summary report. Typically, you use the summary report to identify any discrepancies or problems, and the detail report to uncover more specific information. The workgroup queue report only has a summary report.

- **User Activity Summary:** Summarizes all calls for each user.
- **User Activity Detail:** Lists every call for each user.
- **Trunk Activity Summary:** Summarizes all calls for each trunk.
- **Trunk Activity Detail:** Lists every call for each trunk.
- **Workgroup Agent Summary:** Summarizes all inbound workgroup calls for each agent.
- **Workgroup Agent Detail:** Lists every inbound workgroup call for each agent and optionally, outbound calls. Non-workgroup calls for the agent are also reported.
- **Workgroup Queue Summary:** Summarizes queue activity for every workgroup, including calls that went directly to agents.
- **Workgroup Service Level Summary:** Summarizes data on call processing by the workgroup server.
- **Media Stream Summary:** Summarizes media stream traffic and call quality for calls made over the WAN in multi-site deployments.
- **Media Stream Detail:** Lists media stream made over the WAN in multi-site deployments.
- **Account Code Summary:** Summarizes call information for each account; counts of calls each day, along with their total and average duration. There are also totals for the reporting period.
- **Account Code Detail:** Provides a detailed list of calls that occurred for each account. For each call the date/time of the call, number dialed, the extension making the call and the duration of the call is included. For each account, a

summary is provided of the number of calls, along with their total and average duration.

C.3.1 TMS-CDR Media Stream Statistics

The TMS-CDR Media Stream Statistics feature offers a method of formatting and storing Call Detail Records (CDR) data on media streams and stores that formatted information into a log file on the system, making it easier for the ShoreTel 6.1 system to integrate with various third-party SNMP monitoring tools, and enabling users to acquire a more accurate picture of the traffic patterns in their network. This information can be useful in performing load analysis, identifying peak traffic times, and assisting the customer in setting up competitive pricing strategies.

The system processes media statistics for all calls and formats the raw data into separate lines, with each line partitioned into several columns separated by a comma. Formatted data is then saved in a text file and is subjected to appropriate rollovers similar to the other ShoreWare server logs.

One media stream statistic record will be generated for each RTP stream on a call. Thus, a 3-way fully-meshed conference call would generate 6 records.

C.3.1.1 Formatting

Media statistics are collected and deposited line by line into a file. A delimiter separates one column from the previous one, with no delimiter prior to the first column and none after the last column. The column values will be left-justified and padded with spaces to the right. A value that exceeds the fixed-width column limit will be truncated so that it fits within the limit.

Each line will look similar to the line below:

value-1,value-2,value-3,.....,value-n

The table below summarizes the details of the individual columns:

Column number	Type	Width	Description
1	Integer	20	ID of the line in decimal
2	String	20	Extension Number For anonymous calls, extension number is not available. In such cases, an empty string will be placed at this column.
3	String	16	Name of Extension or Trunk or Phone (UTF-8)

Table C-3 CDR Media Stream Statistics Formatting

Column number	Type	Width	Description
4	Integer	2	Party type in decimal 0 Unknown 1 Station 2 Trunk 3 Virtual 4 Workgroup 5 AutoAttendant 6 VMForward 7 VMLogin 8 BackupAA 9 Anonymous Phone 10 Nightbell 11 Paging 12 Workgroup Agent 13 Unknown 14 RoutePoint 15 ACC 16 HuntGroups 17 GroupPaging
5	String	32	SIP Call ID
6	String	16	Local IP Address (Switch, Trunk Switch, or IP Phone etc.) in dotted decimal form
7	String	16	Remote IP Address (Remote end point. Switch or Trunk or IP Phone etc.) in dotted decimal form
8	Integer	20	Local Site ID (Site ID of extension or trunk or phone that generated starts)
9	String	16	Local Site Name (UTF-8)
10	Integer	3	Code Type 0 Unknown 1 ALAW 2 MULAW 3 Linear – 8 MHz. 4 ADPCM 5 G-729-A 6 G-729-B
11	Integer	10	Payload size (in milliseconds)
12	Integer	2	Status code 0 – Normal 1 - Failure
13	String	12	Starting time of the collection in string HH:MM:SS.MSEC format
14	Integer	20	Duration (in seconds) of the collection in decimal.
15	Integer	20	Number of received packets
16	Integer	20	Number of lost packets
17	Integer	20	Max jitter
18	Integer	20	Underruns
19	Integer	20	Overruns

Table C-3 CDR Media Stream Statistics Formatting

This feature is applicable only on the main (i.e. headquarter) ShoreTel server, and is disabled by default. Enabling the TMS-CDR Media Stream Statistics feature requires making the appropriate changes to the registry settings.

WARNING Do not make any changes to the registry settings unless you are certain you know what you are doing!

Step 1 Click **Start** and select **Run**.

Step 2 Select the **regedit** application to display a window similar to the one shown below:

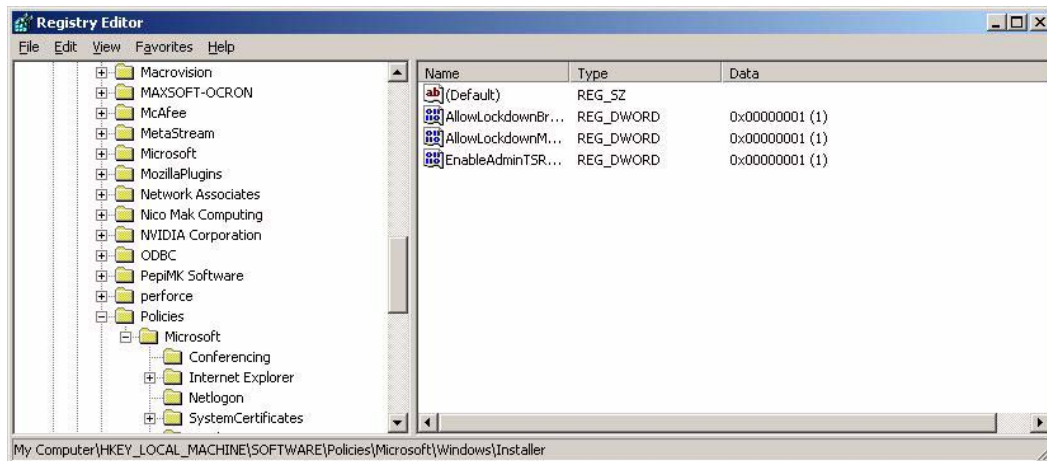


Figure C-3 Registry Editor window

Step 3 Navigate to *SOFTWARE\Shoreline Teleworks\Call Accounting*

Step 4 Double-click the file named *LogMediaStatsToFile* to open the following window:

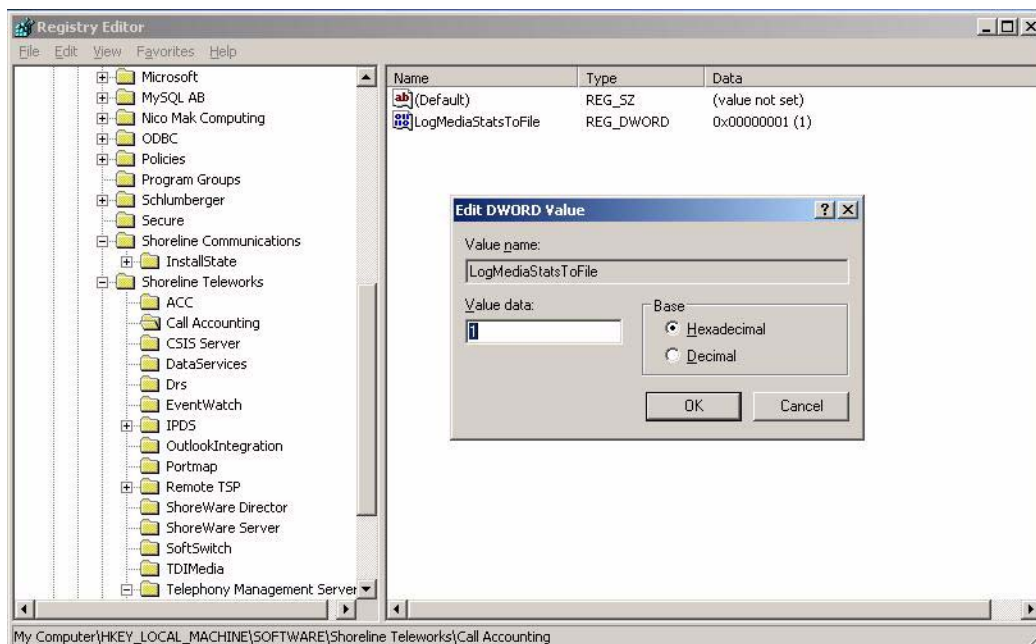


Figure C-4 Value data field

Step 5 Enter **1** in the Value data field and click **OK** to store your changes and enable this feature.

C.3.2 Generating CDR Reports

Refer to Section 20.2 on page 384 for instructions on generating CDR Reports.

C.4 Interpreting CDR Reports

This section provides information on interpreting each of the 12 CDR reports.

C.4.1 User Activity Summary Report

The User Activity Summary Report (Figure C-5) shows a summary of all inbound and outbound calls for each user. This includes the type of calls made, as well as the total duration for all calls. This report can be run as an interval report in which user activity is subtotaled by the selected interval. Additionally, the summary can be run for selected extensions.

Workgroup Service Level Summary Report								
Start: 2/27/2007			End: 2/27/2007					
Start: 7:00:00AM			End: 11:59:59PM			Max. Wait Time: 180 seconds		
Wait Time (sec.)	Abandoned	Handled By Agent	Handled By Voicemail	Transfer	Interflow/Overflow	Handled By Others	Total Calls	
February 27, 2007								
WG all calls answered by agents (x60102)								
	60 - 89	4,050	0	0	0	0	0	4,050
Sub Total	Avg. 00:01:05	4,050	0	0	0	0	0	4,050
WG Overdrive condition (x60101)								
	0 - 29	0	838	0	0	0	0	838
	30 - 59	0	2,012	0	0	0	2	2,014
	60 - 89	5,496	489	0	0	0	32	6,017
	90 - 119	1	0	0	0	0	0	1
	>=180	66	1	0	0	0	0	67
Sub Total	Avg. 00:00:59	5,563	3,340	0	0	0	34	8,937
Sub Total		00:01:01	9,613	3,340	0	0	34	12,987
Total		00:01:01	9,613	3,340	0	0	34	12,987

Figure C-5 User Activity Summary Report

C.4.1.1 Calls Included

The User Activity Summary Report always displays External Calls and can be configured to display Internal Calls. External calls are those calls where the record in the Call table has a CallType of 2 (Inbound) or 3 (Outbound).

C.4.1.2 Name (Extension Field)

For outbound calls, the Name (Extension) field of the Call record always reports the party that initiated the call.

Inbound calls are reported according to the last party involved in the call (excluding voice mail and the auto-attendant). For example, if a call to extension 320 is not answered and the user's Call Handling Mode (CHM) forwards the call to the assistant at extension 452—who answers the call—the Extension field in the Call record contains 452.

When an inbound call is forwarded to voice mail, the Name (Extension) field records the party involved in the call before it was forwarded to voice mail. For example, if a user with extension 320 doesn't answer a call and his or her Call Handling Mode (CHM) forwards the call to voice mail, the extension field is set to 320.

The User Activity Summary Report is described as follows:

Field	Presence/Frequency	Description
Name (Extension)	Once for each extension reported.	The name of the user, last name first. Users without a last name are presented first. Non-users such as Workgroups, Voice Mail, Voice Mail Login, and Auto-attendant are included in the report. The names for these extensions are reported for calls that only interact with these extensions (not a user extension).
Inbound All - Qty, Duration, Average Duration	Once for each period reported.	The quantity, total duration, and average duration for inbound calls during the reporting period are presented. A call is considered inbound if the CallType field of the Call table record is set to 2 (Inbound). If the report is run with intervals, the call is only reported for the interval in which it started, even if it ends in a different interval. The StartTime field in the Call table is used to determine when the call started. Duration represents that extensions time on the call. This is found in the Connect table record's Duration field for this connection (where the Connect.CallTableID matches the Call.ID and Connect.PartyID matches Call.Extension). Since a call is reported during the period in which it starts, but may end during another interval, the duration can be longer than the 30-minute interval period—the total call duration time is reported during the interval in which the call begins. Total Duration during any period is the sum of the duration for the Inbound calls during the period. Average duration is found by dividing this total by the number of calls during the period.
Outbound All - Qty, Duration, Average Duration	Once for each period reported.	The quantity, total duration, and average duration for Outbound calls during the reporting period are presented. A call is considered outbound if the CallType field of the Call table record is set to 3 (Outbound). Duration is calculated here in the same manner as for Inbound calls. Please see that description for details.
Total All - Qty	Once for each period reported.	The quantity, total duration, and average duration of all calls during this period. This simply represents both the Inbound and Outbound columns. Inbound and Outbound quantity and total duration are added together and then averaged.
Outbound Non-Local Trunk - Qty	Once for each period reported.	The quantity, total duration, and average duration for Outbound non-local calls during the reporting period are presented. A call is considered outbound if the CallType field of the Call table record is set to 3 (Outbound). A call is considered Non-Local if the LongDistance field in the Call table is set to true. This flag indicates whether or not the call was long distance from the perspective of the trunk that was used for the call. The calls reported here, are a subset of the calls reported under Outbound all. Duration is calculated in the same manner as for Inbound calls. Please see that description for details.
Outbound WAN Trunk- Qty	Once for each period reported.	The quantity, total duration, and average duration for Outbound non-local calls during the reporting period are presented. A call is considered outbound if the CallType field of the Call table record is set to 3 (Outbound). A call is considered a WAN call if a media stream was established between 2 sites. This is determined by looking in the MediaStream table for any records of media stream for this call (the MediaStream CallID will equal the CallID in the Call table record for the call). The calls reported here, are a subset of the calls reported under Outbound all. Duration is calculated here in the same manner as for Inbound calls. Please see that description for details.
Grand Total		The totals for all the users in the system.

Figure C-6 User Activity Summary Report field descriptions

C.4.2 User Activity Detail Report

The User Activity Detail Report (Figure C-7) shows a list of every call for each user by user. This includes the time a call was received or made, the number dialed, and the trunk used. This report can be run as an interval report in which user activity is sub-totaled by the selected interval.

User Activity Detail Report									
Start Date: 2/27/2007			End Date: 2/27/2007			Show Internal Calls: False			
Start Time:			End Time:						
BACKUP (x70107)									
Date	Time	In/Out	Workgroup	WAN	Dialed #	Calling #	Trunk	Duration	
02/27/2007	12:01:54AM	In-Ex	Yes	No	60102	+34917730021	s14ch02 (23)	0:01:37	
02/27/2007	12:02:20AM	In-Ex	Yes	No	60102	+34917730029	s14ch02 (31)	0:01:36	
02/27/2007	12:02:26AM	In-Ex	Yes	No	60102	+34917730015	s14ch02 (16)	0:01:30	
02/27/2007	12:03:57AM	In-Ex	No	No	60102	+34917730029	s14ch02 (31)	0:01:18	
02/27/2007	12:04:11AM	In-Ex	No	No	60102	+34917730005	s14ch02 (6)	0:01:17	
02/27/2007	12:04:11AM	In-Ex	No	No	60102	+34917730006	s14ch02 (7)	0:01:18	
02/27/2007	12:03:53AM	In-Ex	Yes	No	60102	+34917730001	s14ch02	0:01:37	
02/27/2007	12:03:54AM	In-Ex	Yes	No	60102	+34917730002	s14ch02 (3)	0:01:38	
02/27/2007	12:03:54AM	In-Ex	Yes	No	60102	+34917730018	s14ch02 (20)	0:01:38	
02/27/2007	12:03:54AM	In-Ex	Yes	No	60102	+34917730003	s14ch02 (4)	0:01:39	
02/27/2007	12:03:54AM	In-Ex	Yes	No	60102	+34917730019	s14ch02 (21)	0:01:39	
02/27/2007	12:03:56AM	In-Ex	Yes	No	60102	+34917730020	s14ch02 (22)	0:01:37	
02/27/2007	12:03:56AM	In-Ex	No	No	60102	+34917730004	s14ch02 (5)	0:01:37	
02/27/2007	12:03:57AM	In-Ex	No	No	60102	+34917730015	s14ch02 (16)	0:01:38	
02/27/2007	12:04:14AM	In-Ex	No	No	60102	+34917730007	s14ch02 (8)	0:01:16	
02/27/2007	12:04:15AM	In-Ex	No	No	60102	+34917730008	s14ch02 (9)	0:01:33	
02/27/2007	12:04:47AM	In-Ex	No	No	60102	+34917730012	s14ch02 (13)	0:01:01	
02/27/2007	12:04:47AM	In-Ex	No	No	60102	+34917730027	s14ch02 (29)	0:01:02	
02/27/2007	12:04:48AM	In-Ex	No	No	60102	+34917730013	s14ch02 (14)	0:01:01	
02/27/2007	12:04:48AM	In-Ex	No	No	60102	+34917730028	s14ch02 (30)	0:01:00	
02/27/2007	12:04:48AM	In-Ex	No	No	60102	+34917730014	s14ch02 (15)	0:01:01	
02/27/2007	12:04:52AM	In-Ex	No	No	60102	+34917730021	s14ch02 (23)	0:00:57	
02/27/2007	12:04:50AM	In-Ex	No	No	60102	+34917730009	s14ch02 (10)	0:01:04	
02/27/2007	12:04:50AM	In-Ex	No	No	60102	+34917730022	s14ch02 (24)	0:01:05	
02/27/2007	12:04:50AM	In-Ex	No	No	60102	+34917730023	s14ch02 (25)	0:01:06	
02/27/2007	12:04:50AM	In-Ex	No	No	60102	+34917730024	s14ch02 (26)	0:01:06	
02/27/2007	12:04:50AM	In-Ex	No	No	60102	+34917730010	s14ch02 (11)	0:01:06	
02/27/2007	12:04:50AM	In-Ex	No	No	60102	+34917730025	s14ch02 (27)	0:01:06	
02/27/2007	12:04:50AM	In-Ex	No	No	60102	+34917730026	s14ch02 (28)	0:01:06	
02/27/2007	12:04:50AM	In-Ex	No	No	60102	+34917730011	s14ch02 (12)	0:01:06	
Total							30 Call(s)	0:39:15 Total	0:01:18 Average

Figure C-7 User Activity Detail Report

C.4.2.1 Calls Included

The User Activity Detail Report always displays External Calls and can be configured to display Internal Calls. External calls are those calls where the record in the Call table has a CallType of 2 (Inbound) or 3 (Outbound). All specified calls that have at least one leg with a TalkTime greater than zero is included in the report.

C.4.2.2 Name (Extension Field)

For outbound calls, the Name (Extension) field of the Call record always reports the party that initiated the call.

Inbound calls are reported according to the last party involved in the call (excluding voice mail and the auto-attendant). For example, if a call to extension 320 is not answered, and the user's Call Handling Mode (CHM) forwards the call to his or her assistant at extension 452, who answers the call, the Extension field in the Call record contains 452.

When an inbound call is forwarded to voice mail, the Name (Extension) field records the party involved in the call before it was forwarded to voice mail. For example, if a user with extension 320 doesn't answer a call and his or her Call Handling Mode (CHM) forwards the call to voice mail, the extension field is set to 320.

The User Activity Detail Report is described as follows:

Field	Presence/Frequency	Description
Name (Extension)	Once for each extension reported.	The Last Name, First Name, and Extension being reported upon. These come from the PartyIDLastName, PartyIDName, and PartyID for the Connect record that matches the extension in the Call table. Non-users such as Workgroups, Voice Mail, Voice Mail Login, and Auto-attendant are included in the report. The names for these extensions are reported for calls that only interact with these extensions (not a user extension).
Date/Time	Once for each call reported.	The date and time when the call being reported started. These fields come from the StartTime field in the Call table record for the call being reported. When interval reports are generated, the actual time the call started is reported even if the call continues into another interval.
In/Out	Once for each call reported.	If the CallType field of the Call record for the call is 2 (Inbound), “In-Int” is shown for internal calls and “In-Ext” for external calls. If the CallType is 3 (Outbound), “Out-Int” is shown for internal calls and “Out-Ext” is shown for external calls.
Workgroup	Once for each call reported.	The Workgroup field of the Call record for the call is examined. “Yes” or “No” is displayed depending upon whether or not the field indicates a workgroup call.
WAN	Once for each call reported.	A call is considered a WAN call if a media stream was established between 2 sites. This is determined by looking in the MediaStream table for any media stream records for this call. The MediaStream CallID will equal the CallID in the Call table record for the call. “Yes” or “No” is displayed as appropriate based upon the CDR data.
Dialed #	Once for each call reported.	For outbound calls, this is the number the user dialed and is reported in full, canonical format (including country code). For inbound calls, this is the destination of the call. If the call was a DID or DNIS call, this is the DID or DNIS information for the number dialed. For other types of calls, this is the extension where the call first terminates. The dialed number is retrieved from the Dialed Number field of the Call table record for the call.
Calling #	Once for each call reported.	For inbound calls, this is the calling number—ANI or Caller ID—received by the ShoreTel system and is reported as delivered by the PSTN (may or may not include the 1 in front of the area code). The dialed number is retrieved from the CallerID field of the Call table record. For outbound calls, this is the extension of the user that placed the call. In the case of Outbound calls, this data is retrieved from the PartyID field of the Connect record for the party that initiated the call.
Trunk	Once for each call reported.	This is the first trunk that was used for the call. This data is retrieved from the PortName field of the Connect record for the trunk’s involvement in the call.

Table C-4 User Activity Detail Report field descriptions

Field	Presence/Frequency	Description
Duration		<p>The duration of the call at the indicated extension.</p> <p>For an inbound call, the duration of the call begins when the telephone rings. The duration ends when the user hangs up or when the external party hangs up and disconnect supervision is received by the ShoreTel system.</p> <p>For an outbound call, the duration of the call begins when the user completes dialing. The duration ends when the user hangs up or when the external party hangs up and disconnect supervision is received by the ShoreTel system.</p> <p>This data is retrieved from the Duration field of the Connect table's record for this connection (where the Connect.CallTableID matches the Call.ID and Connect.PartyID matches Call.Extension).</p>
Total		The total number of calls, total duration, and average duration for the user.
Grand Total		The total number of calls, total duration, and average duration for all users.

Table C-4 User Activity Detail Report field descriptions

C.4.3 Trunk Activity Summary Report

The Trunk Activity Summary Report (Figure C-8) shows a summary of all calls for each trunk by trunk group. This includes the type of calls, durations, and average durations made on each trunk.

Trunk Activity Summary Report									
Starting: 4/10/2003 8:02:31 AM					Ending: 4/18/2003 11:02:31 PM				
DID 2Way 1	Inbound			Outbound			Total		
	Qty	Duration	Average Duration	Qty	Duration	Average Duration	Qty	Duration	Average Duration
Most Used	0	0:00:00	0:00:00	10	0:01:51	0:00:11	10	0:01:51	0:00:11
sSch13	3	0:00:20	0:00:06	0	0:00:00	0:00:00	3	0:00:20	0:00:06
sSch14	12	0:00:05	0:00:00	19	0:05:40	0:00:17	31	0:05:45	0:00:11
sSch15	3	0:00:59	0:00:19	34	0:57:18	0:00:40	37	0:58:17	0:00:40
Total	18	0:01:24	0:00:04	113	1:04:49	0:00:34	131	1:06:13	0:00:30
DID 2Way 1	Inbound			Outbound			Total		
	Qty	Duration	Average Duration	Qty	Duration	Average Duration	Qty	Duration	Average Duration
sSch13	1	16:41:52	16:41:52	0	0:00:00	0:00:00	1	16:41:52	16:41:52
sSch15	2	0:00:16	0:00:08	14	16:52:56	1:12:21	16	16:53:12	1:03:19
Total	3	16:42:08	5:34:02	14	16:52:56	1:12:21	17	33:35:04	1:58:32
DID 2Way 2	Inbound			Outbound			Total		
	Qty	Duration	Average Duration	Qty	Duration	Average Duration	Qty	Duration	Average Duration
sSch16	8	0:02:55	0:00:21	0	0:00:00	0:00:00	8	0:02:55	0:00:21
Total	8	0:02:55	0:00:21	0	0:00:00	0:00:00	8	0:02:55	0:00:21
DID 2Way 3	Inbound			Outbound			Total		
	Qty	Duration	Average Duration	Qty	Duration	Average Duration	Qty	Duration	Average Duration
sSch19	0	0:00:00	0:00:00	1	0:00:04	0:00:04	1	0:00:04	0:00:04
Total	0	0:00:00	0:00:00	1	0:00:04	0:00:04	1	0:00:04	0:00:04

Figure C-8 Trunk Activity Summary Report

C.4.3.1 Calls Included

Any trunk activity is reported in the Trunk Activity Summary Report. These calls always have Call records with CallType of 2 (Inbound), 3 (Outbound), or 4 (Tandem). However, with ShoreTel, Release 2.0 and greater, the report is based on the TrunkDirection field in the Connect records representing trunk usage, not the CallType record. There is a 30-day change-over period for upgrades, during which the report reflects both methods of collecting this data.

The Trunk Activity Summary Report is described as follows:

Field	Presence/Frequency	Description
Trunk Group Name	Once for each TrunkGroup being reported upon.	This is the name of the TrunkGroup being reported upon. It's retrieved from the GroupName field of the Connect record representing a trunk's involvement in a call.
Trunk Name	Once for each trunk being reported upon.	The name of the specific trunk used for a call. This is retrieved from the PortName field of the Connect record representing a trunk's involvement in a call.
Inbound Qty, Duration, and Average Duration	Once for each trunk being reported upon.	<p>The quantity, total duration, and average duration for all inbound trunk activity for this trunk during the reporting period are presented.</p> <p>Trunk activity is considered inbound if the TrunkDirection field in the Connect record is set to 2 (Inbound).</p> <p>The quantity is simply a count of the Connect table records during the reporting period for this trunk that indicate inbound trunk usage.</p> <p>Duration is the sum of all the Duration fields for the Connect table records during the reporting period for this trunk that indicate inbound trunk usage.</p> <p>The average duration is found by dividing this total by the quantity reported here.</p>
Outbound Qty, Duration, and Average Duration	Once for each trunk being reported upon.	<p>The quantity, total duration, and average duration for all outbound trunk activity for this trunk during the reporting period are presented.</p> <p>Trunk activity is considered outbound if the TrunkDirection field in the Connect record is set to 3 (Outbound).</p> <p>The quantity is simply a count of the Connect table records during the reporting period for this trunk that indicate outbound trunk usage.</p> <p>Duration is the sum of all the Duration fields for the Connect table records during the reporting period for this trunk that indicate outbound trunk usage.</p> <p>Average duration is found by dividing this total by the quantity reported here.</p>
Total Qty	Once for each trunk being reported upon.	The total calls for the trunk.
Duration	Once for each trunk being reported upon.	The total duration of calls, in hours, minutes, and seconds.
Average Duration	Once for each trunk being reported upon.	The average duration of calls, in hours, minutes, and seconds.
Total		The totals for all trunks in the trunk group.
Grand Total		The totals for all trunks in the system.

Table C-5 Trunk Activity Summary Report field descriptions

C.4.4 Trunk Activity Detail Report

The Trunk Activity Detail Report (Figure C-9) shows a list of every call for each trunk by trunk group. This includes the date and time, the number dialed, and the user's name.

C.4.4.1 Calls Included

Any trunk activity is reported in the Trunk Activity Detail Report. These calls always have Call records with CallType of 2 (Inbound), 3 (Outbound), or 4 (Tandem). However, with ShoreTel Release 2.0 and greater, the report is based on the TrunkDirection field in the Connect records representing trunk usage, not the CallType record. There is a 30-day crossover period for upgrades during which the report will reflect both methods of collecting this data.

Trunk Activity Detail Report										
Starting: 4/10/2003 8:03:46AM				Ending: 4/18/2003 11:03:46PM						
DID 2way 1	Date	Time	In/Out	Diske#	Calling #	User	Duration			
Most Used	04/16/2003	9:43:02AM	Out	+14087671234	303	O/HQ 2	0:00:04			
	04/16/2003	10:01:46AM	Out	+14087671234	303	O/HQ 2	0:00:45			
	04/16/2003	11:47:32AM	Out	+14083313861	303	O/HQ 2	0:00:07			
	04/16/2003	11:52:58AM	Out	+14087671234	303	O/HQ 2	0:00:04			
	04/16/2003	11:53:14AM	Out	+14083313861	303	O/HQ 2	0:00:07			
	04/16/2003	12:25:37PM	Out	+14087671234	202	RM IP1	0:00:09			
	04/16/2003	12:25:56PM	Out	+14087671234	302	RM 1	0:00:10			
	04/16/2003	12:26:44PM	Out	+14087671234	321	Joan Rivers	0:00:11			
	04/16/2003	12:28:06PM	Out	+14087671234	330	Dun No	0:00:09			
	04/16/2003	1:12:31PM	Out	+14087671234	331	Hanz Blitz	0:00:05			
	Sub-Total				10	Call(s)	0:00:11	Average	0:01:51	Total
s5chl3	04/12/2003	4:49:53PM	In	14083313914	+15106564784	Auto-Attendant	0:00:05			
	04/12/2003	6:23:54PM	In	14083313914	+15108937617	Auto-Attendant	0:00:07			
	04/14/2003	3:30:11PM	In	14083313913	+14087301683	Auto-Attendant	0:00:08			
	Sub-Total				3	Call(s)	0:00:06	Average	0:00:20	Total
s5chl4	04/10/2003	9:39:29AM	Out	+14083313913	312	Test User	0:00:27			
	04/10/2003	9:40:44AM	Out	+14083313913	312	Test User	0:00:14			
	04/10/2003	9:40:59AM	In	+14083313913	312	Test User	0:00:00			
	04/10/2003	9:42:05AM	Out	+14083313913	312	Test User	0:00:24			
	04/10/2003	9:42:29AM	In	+14083313913	312	Test User	0:00:01			
	04/10/2003	9:43:51AM	Out	+14083313913	312	Test User	0:00:27			
	04/10/2003	9:44:19AM	In	+14083313913	312	Test User	0:00:01			
	04/10/2003	11:41:55AM	Out	+14083313913	312	Test User	0:00:16			
	04/10/2003	11:42:12AM	In	+14083313913	312	Test User	0:00:00			
	04/10/2003	11:42:52AM	Out	+14083313913	312	Test User	0:00:13			
	04/10/2003	11:43:06AM	In	+14083313913	312	Test User	0:00:01			
	04/10/2003	11:44:11AM	Out	+14083313913	312	Test User	0:00:21			
	04/10/2003	11:44:53AM	In	+14083313913	312	Test User	0:00:00			
	04/10/2003	11:45:25AM	Out	+14083313913	312	Test User	0:00:15			
	04/10/2003	11:45:41AM	In	+14083313913	312	Test User	0:00:00			
	04/10/2003	11:46:33AM	Out	+14083313913	312	Test User	0:00:40			
	04/10/2003	11:47:13AM	In	+14083313913	312	Test User	0:00:00			
	04/10/2003	11:47:59AM	Out	+14083313913	312	Test User	0:00:15			
	04/10/2003	11:48:14AM	In	+14083313913	312	Test User	0:00:01			
	04/10/2003	11:50:25AM	Out	+14083313913	312	Test User	0:00:16			

Figure C-9 Trunk Activity Detail Report

The Trunk Activity Detail Report is described as follows:

Field	Presence/Frequency	Description
Trunk Group Name	Once for each TrunkGroup being reported upon. If the trunk activity for a TrunkGroup requires more than one page to report, then the name is repeated at the top of each additional page.	This is the name of the TrunkGroup being reported upon. This data is retrieved from the GroupName field of the Connect record representing a trunk's involvement in a call.
Trunk Name	Once for each TrunkGroup being reported upon. If the trunk activity for a TrunkGroup requires more than one page to report, then the name is repeated at the top of each additional page.	The name of the specific trunk used. This data is retrieved from the PortName field of the Connect record representing a trunk's involvement in a call.
Date	Once for each trunk activity in the report.	The date is extracted from the ConnectTime field in the Connect record representing the trunk's involvement in the call. This is the date the trunk was added to the call.
Time	Once for each trunk activity in the report.	The time is extracted from the ConnectTime field in the Connect record representing the trunk's involvement in the call. This is the time the trunk was added to the call.
In/Out	Once for each trunk activity in the report.	Trunk activity is considered "In" if the TrunkDirection field in the Connect record is set to 2 (Inbound), otherwise it is considered "Out."

Table C-6 Trunk Activity Detail Report field descriptions

Field	Presence/Frequency	Description
Dialed #	Once for each trunk activity in the report.	<p>For outbound calls, this is the number the user dialed and is reported in full, canonical format (including country code, etc.). For an inbound call this is the destination of the call. If the call was a DID or DNIS call, this is the DID or DNIS information for the number dialed. For other types of calls, this is the extension where the call first terminates.</p> <p>For inbound calls (CallType = 2 in Call record) this data is retrieved from the DialedNumber field in the Call record. For other calls, it is retrieved from the PartyId field of the Connect record for the trunk activity.</p> <p>Inbound and outbound are relative to the call, not trunk usage.</p>
Calling #	Once for each trunk activity in the report.	<p>For inbound calls, this is the calling number—ANI or Caller ID—received by the ShoreTel system and is reported as delivered by the PSTN (may or may not include the 1 in front of the area code). For outbound calls, this is the extension of the user that placed the call.</p> <p>For outbound calls (CallType = 3 in Call record) this data is retrieved from the Extension field in the Call record. For other types of calls, it is retrieved from the CallerID field in the Call table.</p> <p>Inbound and outbound are relative to the call, not trunk usage.</p>
User	Once for each trunk activity in the report.	<p>The name associated with the extension that was the initial target of the call. For outbound calls (CallType = 3 in Call record), the user is the extension that first initiated the call. For inbound calls (CallType = 2 in Call record), the user is the extension that was the initial target of the call. In the case of Tandem calls (CallType = 4 in the Call record), nothing is shown.</p> <p>This data is retrieved from the PartyIDName and PartyIDLastName fields of the Connect record for the party that initiated the call (ConnectReason = 19, “Originate” for an outbound call) or was the target of the call (ConnectReason = 17, “Called” for inbound call).</p> <p>Inbound and outbound are relative to the call, not trunk usage.</p>
Duration	Once for each trunk activity in the report.	<p>The duration of the trunk activity. This data is retrieved from the Duration field of the Connect record for the trunk's involvement in the call.</p> <p>For an inbound call, the duration of the call begins when the trunk is seized and includes the ring time, talk time, and hold time. The duration ends when the user hangs up or when the external party hangs up and disconnect supervision is received by the ShoreTel system.</p> <p>For an outbound call, the duration of the call begins when the trunk is seized. The duration ends when the user hangs up, or when the external party hangs up and disconnect supervision is received by the ShoreTel system.</p>
Subtotal		The total of calls for the trunk.
Total		The total of calls for the trunk group.
Grand Total		The total of calls for all trunks in the system.

Table C-6 Trunk Activity Detail Report field descriptions

C.4.5 Workgroup Agent Summary Report

The Workgroup Agent Summary Report (Figure C-10) shows a summary of inbound workgroup calls and agent activity by the workgroup.

Workgroup Agent Summary Report													
Start: 4/10/2003			End: 4/18/2003			Duration Format: hh:mm:ss							
Start: 8:50:57AM			End: 10:50:57PM										
Inbound Workgroup Calls			Inbound User Calls			Outbound Calls			Agent Activity				
Qty	Duration	Average Duration	Qty	Duration	Average Duration	Qty	Duration	Average Duration	Qty	Duration	Average Duration	Total Login	
Mix HQ Remote (s400)													
1, HQ (s301)													
4/14/2003 (Mon)	0	00:00:00	00:00:00	0	00:00:00	00:00:00	1	00:00:08	00:00:08	1	00:00:08	00:00:08	00:00:00
4/16/2003 (Wed)	0	00:00:00	00:00:00	2	16:41:54	08:20:57	0	00:00:00	00:00:00	2	16:41:54	08:20:57	00:00:00
4/17/2003 (Thu)	0	00:00:00	00:00:00	0	00:00:00	00:00:00	5	00:03:06	00:00:37	5	00:03:06	00:00:37	00:00:00
User Sub Total	0	00:00:00	00:00:00	2	16:41:54	08:20:57	6	00:03:14	00:00:32	8	16:45:08	02:05:38	00:00:00
1, RM (s302)													
4/10/2003 (Thu)	0	00:00:00	00:00:00	1	00:00:04	00:00:04	9	00:01:08	00:00:07	10	00:01:12	00:00:07	00:00:00
4/11/2003 (Fri)	0	00:00:00	00:00:00	0	00:00:00	00:00:00	2	00:00:20	00:00:10	2	00:00:20	00:00:10	00:00:00
User Sub Total	0	00:00:00	00:00:00	1	00:00:04	00:00:04	11	00:01:28	00:00:08	12	00:01:32	00:00:07	00:00:00
Workgroup Sub total	0	00:00:00	00:00:00	3	16:41:58	05:33:59	17	00:04:42	00:00:16	20	16:46:40	00:50:20	00:00:00

Figure C-10 Workgroup Agent Summary Report

C.4.5.1 Calls Included

This report includes calls routed to workgroup agents by the workgroup server, and non-workgroup calls (both inbound and outbound). The report assigns non-workgroup calls to an agent's membership within a workgroup by examining the workgroup the agent was logged into during or before the call. No calls are reported when an agent is logged out. You can find Agent logins by examining the AgentActivity table for records with State = 5 (LogIn/Out).

Workgroup agents can be a member of more than one workgroup. When they log in, their login time is reported for all workgroups of which they are a member.

Non-workgroup calls are reported against the workgroup with the lowest dial number that the agent is a member of when the call is made. For example, if the agent is a member of workgroups with dial numbers of 1100, 1200, and 1250, non-workgroup calls are reported against 1100.

The StartTimeStamp field in these Agent Activity records represents the time that an agent logged into a specific workgroup (identified by the WorkgroupDN and WorkgroupName fields). The EndTimeStamp field records the time the agent logged out (this can be null when the agent is still logged into the workgroup).

This report is call centric. While it does report agent activity, which consists of agent wrap-up and login time, the report will only show this information for periods during which there was a call for the agent (workgroup or non-workgroup).

The Workgroup Agent Summary Report is described as follows:

C.4.6 Workgroup Agent Detail Report

The Workgroup Agent Detail Report (Figure C-11) shows a list of every call for each agent by workgroup.

An agent appears in the Workgroup Agent Detail Report if the agent had any workgroup call activity during the reporting period.

Workgroup Agent Detail Report									
Start: 4/10/2003		8:53:31AM		End: 4/10/2003		10:53:31PM			
Date/Time	Dialed #	Calling #	Call Type	Trunk	Call Duration	Wrap up Duration	Queue Duration	Total Duration	
April 10, 2003									
Mix HQ Remote x(400)									
1, RM (x302)									
4/10/2003 10:46:58AM	312	302	Out		00:00:05	00:00:00	00:00:00	00:00:05	
4/10/2003 1:58:47PM	202	302	Out		00:00:07	00:00:00	00:00:00	00:00:07	
4/10/2003 2:47:47PM	+14083313300	302	Out	s5chl5	00:00:05	00:00:00	00:00:00	00:00:05	
4/10/2003 3:02:57PM	202	302	Out		00:00:07	00:00:00	00:00:00	00:00:07	
4/10/2003 3:03:14PM	202	302	Out		00:00:11	00:00:00	00:00:00	00:00:11	
4/10/2003 3:04:33PM	+14083313300	302	Out	s5chl5	00:00:05	00:00:00	00:00:00	00:00:05	
4/10/2003 3:04:55PM	202	302	Out		00:00:02	00:00:00	00:00:00	00:00:02	
4/10/2003 3:04:52PM	202	303	In		00:00:04	00:00:00	00:00:00	00:00:04	
4/10/2003 3:16:32PM	+16504837757	302	Out	s5chl5	00:00:19	00:00:00	00:00:00	00:00:19	
4/10/2003 5:08:37PM	+14087671212	302	Out	s5chl5	00:00:07	00:00:00	00:00:00	00:00:07	
Sub Total			10 Calls	00:00:07	Average	00:01:12	00:00:00	00:00:00	00:01:12
Sub Total			10 Calls	00:00:07	Average	00:01:12	00:00:00	00:00:00	00:01:12
Sales x(801)									
IP1, HQ (x201)									
4/10/2003 1:56:19PM	201	312	In		00:00:13	00:00:00	00:00:00	00:00:13	
4/10/2003 1:58:19PM	201	202	In		00:00:18	00:00:00	00:00:00	00:00:18	
Sub Total			2 Calls	00:00:15	Average	00:00:31	00:00:00	00:00:00	00:00:31
Sub Total			2 Calls	00:00:15	Average	00:00:31	00:00:00	00:00:00	00:00:31
Support x(800)									
2, HQ (x303)									
4/10/2003 9:40:30AM	14083313913	Unknown	InWG	s5chl9	00:00:21	00:00:30	00:00:02	00:00:53	
4/10/2003 9:41:48AM	14083313913	Unknown	InWG	s5chl9	00:00:34	00:00:30	00:00:02	00:01:06	
4/10/2003 9:43:29AM	14083313913	Unknown	InWG	s5chl9	00:00:38	00:00:30	00:00:07	00:01:15	
4/10/2003 9:46:18AM	14083313913	Unknown	InWG	s5chl9	00:00:02	00:00:30	00:00:08	00:00:40	
4/10/2003 11:41:39AM	14083313913	Unknown	InWG	s5chl9	00:00:25	00:00:30	00:00:02	00:00:57	
4/10/2003 11:42:56AM	14083313913	Unknown	InWG	s5chl20	00:00:04	00:00:30	00:00:02	00:00:36	
4/10/2003 11:43:55AM	14083313913	Unknown	InWG	s5chl9	00:00:30	00:00:30	00:00:01	00:01:01	
4/10/2003 11:44:43AM	14083313913	Unknown	InWG	s5chl9	00:00:26	00:00:30	00:00:27	00:01:23	
4/10/2003 11:46:36AM	14083313913	Unknown	InWG	s5chl20	00:00:11	00:00:30	00:00:20	00:01:01	

Figure C-11 Workgroup Agent Detail Report

C.4.6.1 Calls Included

This report includes calls routed to workgroup agents by the workgroup server, and non-workgroup calls (both inbound and outbound). The report assigns non-workgroup calls to an agent's membership within a workgroup by examining the workgroup the agent was logged into during or before the call. Non-workgroup calls made while an agent is logged out are not reported.

Workgroup agents can be a member of more than one workgroup. When they log in, their login time is reported for all workgroups of which they are a member.

Non-workgroup calls are reported against the workgroup with the lowest dial number that the agent is a member of when the call is made. For example, if the agent is a member of workgroups with dial numbers of 1100, 1200, and 1250, non-workgroup calls are reported against 1100.

You can find agent logins by examining the AgentActivity table for records with State = 5 (LogIn/Out). The StartTimeStamp field in these AgentActivity records represents the time that an agent logged into a specific workgroup (identified by the WorkgroupDN and WorkgroupName fields). The EndTimeStamp field records the time the agents logged out (this can be null when the agent is still logged into the workgroup).

The Workgroup Agent Detail Report is described as follows:

C.4.7 Workgroup Service Level Summary Report

The Workgroup Service Level Summary Report provides information related to the workgroup server call processing. Every time the workgroup server processes a call, a record about its disposition is added to the QueueCall table. Generally, this occurs once when the call gets processed by the server. However, in the case of call forwarding, the same call can pass through the workgroup server more than once.

Field	Presence/Frequency	Description
Workgroup Name	Once for each workgroup being reported upon.	The name of the workgroup agent.
Agent Name	Once for each agent reported upon (within an interval if intervals are selected when running the report).	The name of the agent and his or her extension. These come from the AgentActivity table's AgentLastName, AgentFirstName, and AgentDN fields.
Date/Time	Once for each call reported.	The date and time for the call being reported. These fields come from the StartTime field in the Call table record for the reported call. When interval reports are generated, the call is reported for the time when it starts even if it continues into another interval.
Dialed #	Once for each call reported.	For inbound calls, this is the destination of the call. If the call was a DID or DNIS call, this is the DID or DNIS information for the number dialed. For other types of calls, this is the extension where the call first terminates. The number dialed to initiate the call. This comes from the Call record's DialedNumber field.
Calling #	Once for each call reported.	The caller that initiated the call. For inbound calls, this is the calling number—ANI or Caller ID—received by the ShoreTel system and is reported as delivered by the PSTN (may or may not include the “1” in front of the area code). This comes from the Call table's CallerID field.
Call Type	Once for each call reported.	Indicates whether this is an incoming workgroup call (“InWG”), an inbound non-workgroup call (“In”), or an outbound call (“Out”). A call is categorized as an inbound workgroup call if the user joined the call as a workgroup agent. This is determined by examining the Connect record for the user's time on the call. The PartyType in the Connect record must be 12 (Workgroup Agent). A call is categorized as an inbound non-workgroup call if the Call record's CallType = 1 or 2 (internal or inbound), and the user's Connect record has PartyType = 1 (station), and the user was not the originator of the call (ConnectReason in the Connect table not equal to 19-- originate). A call is categorized as outbound if the user originated the call (as shown by the Connect record having ConnectReason = 19). These calls have CallType = 1 or 3 (internal or outbound) in the call record. Note that for all calls, those calls with CallType = 1 (internal) are included only if the option to include internal calls is chosen. Calls that involve multiple legs are also reported as: Transfer—Call was transferred. Conference—Call was conferenced Monitor—Call was monitored Barge-In—Call was barged
Other Calls	Once for each call reported.	Indicates calls that do not belong to Call Type (i.e. those that are no In, InWG, or Out). This includes: Transfer—Call was transferred. Conference—Call was conferenced Monitor—Call was monitored Barge-In—Call was barged
Trunk	Once for each call reported.	This is the first trunk that was used for the call. This data is retrieved from the PortName field of the Connect record for the trunk's involvement in the call. In the case of calls not involving a trunk, it will be blank (this can occur with internal calls).

Table C-7 Workgroup Agent Detail Report field descriptions

Field	Presence/Frequency	Description
Call Duration	Once for each call reported.	The duration of the call. Duration reports the time of the user's involvement in the call. It's reported by summing the TalkTime, RingTime, and HoldTime fields in the Connect record representing involvement in the call. Since a call is reported during the period in which it starts (as identified by the StartTime in the Call table record for the call) but may end during another interval, the duration can be longer than the 30- or 60-minute interval period. The total duration is reported during the interval in which the call begins.
Wrap-Up Duration	Once for each call reported.	Only applicable to inbound workgroup calls. This is the amount of time, if any, the agent spent in wrap-up after completing the call. The Wrap-up Duration is the difference between the StartTimeStamp and EndTimeStamp in the wrap-up record in the AgentActivity table for that agent.
Queue Duration	Once for each call reported.	Only applicable to inbound workgroup calls, the amount of time that the call was in the workgroup queue before it was assigned to the agent. This data is retrieved from the Duration field of the QueueCall table record for the call.
Total Duration	Once for each call reported.	The Total Duration includes the Queue Duration, Call Duration, and Wrap-up Duration. It is generally less than the total time the call spends within the ShoreTel system. The time between when the trunk was seized and the call was accepted by the workgroup, or any time the call spends with a menu or other extension, is not reflected.
Sub-total		The total calls for the workgroup agent.
Total		The total calls for all agents in the workgroup.
Grand Total		The total calls for all agents in the system.

Table C-7 Workgroup Agent Detail Report field descriptions

For example, a call made to the workgroup server is transferred to an extension. If that extension's call handling mode forwards the call to the same or a different workgroup, the call passes through the workgroup server more than once. The rule in these cases is simple—for each time the workgroup server disposes of a call, a record is added to the QueueCall table.

External calls to a workgroup are always included in the report. Internal workgroup calls are only included in the report if the option is taken to include them (the default is not included). The CallType field in the Call table is examined to determine if the call is internal or external. If the CallType is 1 (extension to extension), it is an internal call; otherwise it is an external call. The QueueCall record for a call processed by the workgroup server has a ConnectTableID that identifies the Connect table entry for the workgroup server being added to the call. The Connect table entry has a CallTableID field that is examined to determine the Call table record for the call, whether the call is internal or external.

If the workgroup service is not operational, the call is not processed by the workgroup server (it simply goes to the backup extension). Calls directed to the workgroup but not processed by the service because it is down, are not included in this report. When this occurs, there is no record of the call in the QueueCall table, since records are only added to the table when the workgroup server processes the call.

The Workgroup Service Level Summary Report is described below.

Workgroup Service Level Summary Report								
Start : 2/27/2007			End: 2/27/2007					
Start: 7:00:00AM			End: 11:59:59PM					
Max. Wait Time: 180 seconds								
Wait Time (sec.)	Abandoned	Handled By Agent	Handled By Voicemail	Transfer	Interflow/ Overflow	Handled By Others	Total Calls	
February 27, 2007								
WG all calls answered by agents (x60102)								
	60 - 89	4,050	0	0	0	0	0	4,050
Sub Total	Avg. 00:01:05	4,050	0	0	0	0	0	4,050
WG Overdrive condition (x60101)								
	0 - 29	0	838	0	0	0	0	838
	30 - 59	0	2,012	0	0	0	2	2,014
	60 - 89	5,496	489	0	0	0	32	6,017
	90 - 119	1	0	0	0	0	0	1
	>=180	66	1	0	0	0	0	67
Sub Total	Avg. 00:00:59	5,563	3,340	0	0	0	34	8,937
Sub Total	00:01:01	9,613	3,340	0	0	0	34	12,987
Total	00:01:01	9,613	3,340	0	0	0	34	12,987

Figure C-12 Workgroup Service Level Summary Report

Field	Presence/Frequency	Description
Wait Time (sec.)	Shown once for each 30-second period in workgroup/internal where there are calls to be reported.	Range of wait-for-service times, The wait time is divided into 30-second intervals. Information for the calls is reported for the interval in which it falls, according to when the call moved off the workgroup. The actual wait or service time for each workgroup call is found in the Duration field of the QueueCall table. This duration is the time from when the call is offered to the workgroup server until it leaves the call queue.
Abandoned	Once for each period reported.	Number of callers who abandoned the call (hung up) during the period. Those QueueCall records with the ExitReason set to 7 (Abandoned) are counted as abandoned.
Handled by Agent	Once for each period reported.	Number of calls handled by agents during the period. Those QueueCall records with the ExitReason set to 1 (TransferToAgent) are counted as handled by an agent. A call that is picked up or unparked by an agent that is a member of the same workgroup is also counted as Handled by Agent.
Handled by Voice Mail	Once for each period reported.	Number of calls that went to the workgroup's voice mail (either as a result of call handling, or when the caller chose the transfer to voice mail option). Those QueueCall reports with the TargetType set to 3 (mailbox) and ExitReason set to 2, 3, 4, or 5 (ForwardAlways, ForwardBusy, ForwardNoAnswer, or ForwardNoLoginAgent), or 9 (TransferVM) with the TargetDN field equal to the workgroup DN itself are counted as Handled by Voice Mail.
Transfer	Once for each period reported.	Number of calls transferred by WorkGroup agents.

Table C-8 Workgroup Service Level Summary Report field descriptions

Field	Presence/Frequency	Description
Interflow / Overflow	Once for each period reported.	Number of automatic call transfers, based on caller wait time to a dialable number (interflow) or to another WorkGroup queue (overflow).
Handled by Others	Once for each period reported.	Number of calls handled by others (not workgroup agents or voice mail). Any call for the workgroup that isn't reported as Abandoned, Handled by Agent, or Handled by Voice Mail is counted as Handled by Others. Calls that are picked up by non-agents, or agents that do not belong to the group are counted as Handled by Others.
Total Calls	Once for each period reported.	Sum of Abandoned, Handled by Agent, Handled by Voice Mail, Picked Up from the Queue, Unparked from the Queue, and Handled by Others for the period.

Table C-8 Workgroup Service Level Summary Report field descriptions

C.4.8 Workgroup Queue Summary Report

The Workgroup Queue Summary Report (Figure C-13) shows a summary of queue activity and how the calls interact with the queue. You can run the report with fixed interval sub-totals.

Workgroup Queue Summary Report											
Start: 2/27/2007						End: 2/27/2007					
Start:						End:					
Abandoned	Handled by Agent	Handled by WG Voicemail	Transfer	OverFlow/ Interflow	Handled by Others	Maximum Abandon Time	Average Abandon Time	Maximum Handled Time	Average Handled Time	Total Calls	
WG all calls answered by agents (x60102)											
02/27/2007 (Tue)	15,221	0	0	0	0	00:02:09	00:01:05	00:00:00	00:00:00	15,221	
Sub Total	15,221	0	0	0	0	00:02:09	00:01:05	00:00:00	00:00:00	15,221	
WG Overdrive condition (x60101)											
02/27/2007 (Tue)	16,011	10,409	0	0	99	00:05:11	00:01:05	00:02:09	00:00:38	26,519	
Sub Total	16,011	10,409	0	0	99	00:05:11	00:01:05	00:02:09	00:00:38	26,519	
Total	31,232	10,409	0	0	99	00:05:11	00:01:05	00:02:09	00:00:38	41,740	

Figure C-13 Workgroup Queue Summary Report

C.4.8.1 Calls Included

The key determinant in this report is which workgroup server processes the call. Each time the workgroup server processes a call, a record about the call's disposition is added to the QueueCall table. In most cases the call is recorded just once, but if forwarded, a call can be recorded twice. Normally the call comes in and is processed by the server where it is routed to an agent. The caller then chooses to go to voice mail or another destination, or hangs up (abandons the call) before it is routed beyond the workgroup. Since the report shows how the call was disposed of by the workgroup server, the call is reported once in the report. However, if the call is forwarded, the same call can pass through the workgroup server more than once.

For example, a call is made to the workgroup server. While on the call, the user chooses to transfer to an extension. That extension's call handling mode then forwards the call to the same or a different workgroup. In this case, the call passes through the workgroup server

more than once and is reported on once each time the workgroup server disposes of it. For each time the workgroup server disposes of a call there is a record added to the QueueCall table.

External calls to a workgroup are always included in the report. Internal workgroup calls are only included in the report if the option to include them is enabled (by default they are not). The CallType field in the Call table is examined to determine if the call is internal or external. If the CallType is 1 (extension to extension), it is an internal call; otherwise it is an external call. The QueueCall record for a call processed by the workgroup server has a ConnectTableID that identifies the Connect table entry for the workgroup server being added to the call. The Connect table entry has a CallTableID field that is then examined to determine the Call table record for the call. It is this record's CallType that is examined to determine whether the call is internal or external.

If the workgroup service is not operational, the call is not processed by the workgroup server (it simply goes to the backup extension). These calls are not included in the report. When this occurs, there is no record of the call in the QueueCall table, since records are only added to that table when the workgroup server processes the call.

The Workgroup Queue Summary Report is described as follows:

Field	Presence/Frequency	Description
Workgroup Name		The name of the workgroup.
Calls Abandoned	Once for each period reported.	The number of callers who hung up or otherwise disconnected while waiting in queue. Those QueueCall records with the ExitReason set to 7 (abandoned) are counted as Abandoned.
Calls Handled by Agent	Once for each period reported.	The number of calls that were answered by agents in the workgroup. Those QueueCall records with the ExitReason set to 1 (TransferToAgent) are counted as Handled by Agent.
Calls Handled by Voice Mail	Once for each period reported.	Number of calls that went to the workgroup's voice mail (either as a result of call handling or when the caller chose to transfer to voice mail). Those QueueCall records with the TargetType set to 3 (Mailbox), and ExitReason set to 2, 3, 4, or 5 (ForwardAlways, ForwardBusy, ForwardNoAnswer, or ForwardNoLoginAgent) or 9 (TransferVM) with the TargetDN field equal to the workgroup DN itself are counted as Handled by Voice Mail.
Transfer	Once for each period reported	Number of calls transferred by WorkGroup agents.
Overflow / Interflow	Once for each period reported.	Number of automatic call transfers, based on caller wait time to a dialable number (interflow) or to another WorkGroup queue (overflow).
Calls Handled by Others	Once for each period reported.	Number of calls handled by others (not workgroup's agents or voice mail). Any call for the workgroup that isn't reported as Abandoned, Handled by Agent, Picked Up from the Queue, Unparked from the Queue, or Handled by Voice Mail is counted as Handled by Others.
Maximum Abandoned Time	Once for each period reported.	The maximum time during the period that a caller who abandoned the call stayed on the line. The QueueCall records are examined for calls reported as abandoned during this period. The DurationSeconds field with the largest value for abandoned calls is reported.

Table C-9 Workgroup Queue Summary Report field descriptions

Field	Presence/Frequency	Description
Average Abandoned Time	Once for each period reported.	The average time during the period that those callers who abandoned the call stayed on the line. The sum of the DurationSeconds field in all of the QueueCall records for abandoned calls during the period divided by the number of such calls to report the average time abandoned.
Maximum Handled Time	Once for each period reported.	The maximum time during the period that a caller stayed on the line before the call was handled (by agent, voice mail, or others). Note that the maximum time could be zero even though there were handled calls in the case of the call being forwarded immediately to voice mail. The QueueCall records are examined for calls reported as handled during this period. The DurationSeconds field with the largest value for abandoned calls is reported.
Average Handled Time	Once for each period reported.	The average time during the period that a caller was on the line before the call was handled (by an agent, voice mail, or others). Note that the average time could be zero even though there were handled calls in the case of the call being forwarded immediately to voice mail. The sum of the DurationSeconds field in all of the QueueCall records for handled calls during the period divided by the number of such calls to report the average time abandoned.
Total Calls	Once for each period reported.	All calls passed through the workgroup. This includes calls that go straight to agents without waiting in queue. Sum of Abandoned, Handled by Agent, Handled by Voice Mail, and Handled by Others for the period.

Table C-9 Workgroup Queue Summary Report field descriptions

C.4.9 Media Stream Summary Report

The Media Stream Summary Report (Figure C-14) shows the summary of call quality and call traffic for calls made over the WAN in multi-site deployments. By understanding the amount of time the WAN is used for calls, you can estimate the amount of toll charges your organization is saving. In addition, by understanding the jitter and packet loss, you can get an approximation of the quality of the WAN link and use this to influence your service provider if required.

The Media Stream Report lists a matrix of all sites and the links to other sites on the system. Media streams are reported rather than calls, since this report focuses on the exact amount of bandwidth used. Calls can be quite complex involving multiple parties, including users, voice mail, and auto-attendant. Each media stream that is reported includes the associated Call ID (Call Identification) that can be correlated to the parties on the call for troubleshooting purposes using the CDR database.

Media Stream Summary Report									
Starting: 2/27/2007 12:00:00AM Ending: 2/27/2007 11:59:59PM									
		Quality				All Calls Traffic Volume			
Site A	Site B	Avg Jitter (ms)	Max Jitter (ms)	% Packets Lost	Blocked Calls	Total	Duration	Avg Duration	
HQ	HQ	1.66	6.00	0.00	0	4,819	84:06:03	00:01:02	
HQ	RM1 - DVM - FUJII12	1.84	5.00	0.00	0	1,613	28:15:12	00:01:03	
HQ	RM2_SGE1	2.59	5.00	0.00	0	125	02:17:50	00:01:06	
HQ	RM3 - FUJIT1	2.61	7.00	0.00	0	1,753	24:45:26	00:00:50	
HQ	RM4 - FUJIB	1.95	6.00	0.00	0	1,641	28:45:20	00:01:03	
RM1 - DVM - FUJII12	RM2_SGE1	2.35	5.00	0.00	0	43	00:47:08	00:01:05	
RM1 - DVM - FUJII12	RM3 - FUJIT1	3.26	6.00	0.00	0	114	02:01:04	00:01:03	
RM3 - FUJIT1	RM2_SGE1	2.32	7.00	0.00	0	31	00:21:45	00:00:42	
RM4 - FUJIB	RM2_SGE1	2.81	5.00	0.00	0	36	00:41:34	00:01:09	
RM4 - FUJIB	RM3 - FUJIT1	3.04	6.00	0.00	0	94	01:34:15	00:01:00	

Figure C-14 Media Stream Summary Report

C.4.9.1 Calls Included

This report summarizes media streams (not calls) between the two sites. Media streams can be for extensions or trunks. You can configure the report to display information for all calls or for only intersite calls. IP phone media streams are not included in this report.

The Media Stream Summary Report is described as follows:

Field	Presence/Frequency	Description
Site A	Once for each pair of sites being reported.	The name of the site. This data is retrieved from the ASiteName field in the MediaStream table.
Site B	Once for each pair of sites being reported.	The name of the site that communicates to Site A. This data is retrieved from the BSiteName field in the MediaStream table.
Quality - Avg Jitter	Once for each pair of sites being reported.	The average jitter is the average of the per-media stream maximum jitter between the sites given in milliseconds. This data is retrieved from the A MaxJitter and B MaxJitter fields in the MediaStream table.
Quality - Max Jitter	Once for each pair of sites being reported.	The maximum jitter is the worst jitter encountered on any media stream between the sites given in milliseconds. This data is retrieved from the A MaxJitter and B MaxJitter fields in the MediaStream table. The jitter buffer should be larger than this value for proper operation. The 'max jitter' value in this report is only recorded up to the maximum jitter buffer value configured in Director.
Quality - % Packet Loss	Once for each pair of sites being reported.	This is the number of packets that were expected to arrive but did not arrive at the destination. Lost packets were mostly likely dropped on their way through the network.
Quality - Blocked Calls	Once for each pair of sites being reported.	The number of media that were not routed across the WAN due to insufficient WAN bandwidth (admission control reached). This could indicate that more WAN bandwidth is required. This is a count of the number of records in the MediaStream table between the two sites with FailureCode = 1 (Admission Control Inhibited Call).
Traffic Volume - Total	Once for each pair of sites being reported.	The number of media streams used between the two sites as recorded in the MediaStream table.
Traffic Volume - Duration	Once for each pair of sites being reported.	The duration of all the media streams used between the two sites. The value is the sum of duration for all records between the two sites in the MediaStream table.
Traffic Volume - Avg Duration	Once for each pair of sites being reported.	The average duration of all the media streams used between the two sites. The average is total duration of the media streams between the two sites divided by the number of such media streams.

Table C-10 Media Stream Summary Report field descriptions

C.4.10 Media Stream Detail Report

The Media Stream Detail Report (Figure C-15) shows details of each media stream placed over the WAN. You can configure the report to display information for all calls or for only intersite calls.

C.4.10.1 Call Included

See the Media Stream Summary Report for information about selection. This report calls out each media stream established between two sites.

IP phone media streams are not included in this report.

The Media Stream Detail Report is described as follows:

Media Stream Detail Report								
Starting: 2/27/2007 12:00:00AM				Ending: 2/27/2007 11:59:59PM			All Calls	
Site A	Site B	Start Time	WAN	Call ID	Encoding	Max Jitter (ms)	% Packets Lost	Duration
HQ	RMD_SGE1	2/27/2007 10:27:44AM	Yes	6D42EA2C	G729A	3	0.00	00:01:04
HQ	RMD_SGE1	2/27/2007 9:50:47AM	Yes	8C850D0F	G729A	4	0.00	00:01:04
HQ	RMD_SGE1	2/27/2007 10:12:48AM	Yes	78CF1789	G729A	4	0.00	00:01:04
HQ	RMD_SGE1	2/27/2007 10:27:45AM	Yes	683F10DD	G729A	3	0.00	00:01:04
HQ	RMD_SGE1	2/27/2007 10:12:48AM	Yes	73CB3E3A	G729A	4	0.00	00:01:04
HQ	RMD_SGE1	2/27/2007 9:50:47AM	Yes	827D5A71	G729A	2	0.00	00:01:04
HQ	RMD_SGE1	2/27/2007 10:29:16AM	Yes	F5224007	G729A	0	0.00	00:00:01
HQ	RMD_SGE1	2/27/2007 9:50:48AM	Yes	B9E36619	G729A	1	0.00	00:01:03
HQ	RMD_SGE1	2/27/2007 10:13:44AM	Yes	A62D7093	G729A	4	0.00	00:01:04
HQ	RMD_SGE1	2/27/2007 10:13:58AM	Yes	9721E4A6	G729A	2	0.00	00:01:04
HQ	RMD_SGE1	2/27/2007 9:51:45AM	Yes	AD3400DD	G729A	3	0.00	00:01:04
HQ	RMD_SGE1	2/27/2007 9:52:09AM	Yes	A0D0278E	G729A	3	0.00	00:01:04
HQ	RMD_SGE1	2/27/2007 10:14:09AM	Yes	C88416FF	G729A	4	0.00	00:01:05
HQ	RMD_SGE1	2/27/2007 10:14:09AM	Yes	C4803DB0	G729A	2	0.00	00:01:05
HQ	RMD_SGE1	2/27/2007 9:52:09AM	Yes	F61195CD	G729A	3	0.00	00:01:04
HQ	RMD_SGE1	2/27/2007 10:15:06AM	Yes	FD6A435D	G729A	1	0.00	00:01:05
HQ	RMD_SGE1	2/27/2007 9:52:10AM	Yes	F10DBC7E	G729A	3	0.00	00:01:04
HQ	RMD_SGE1	2/27/2007 9:52:10AM	Yes	EC09E13F	G729A	3	0.00	00:01:04
HQ	RMD_SGE1	2/27/2007 10:15:07AM	Yes	F36290BF	G729A	3	0.00	00:01:04
HQ	RMD_SGE1	2/27/2007 10:15:07AM	Yes	EE5EB770	G729A	1	0.00	00:01:04
HQ - RMD_SGE1 Total						122 Call(s) - Average Duration 00:01:06 - Total Duration 02:14:37		
HQ	RMB - FUJITI	2/27/2007 1:50:38AM	Yes	F5DB9F71	G729A	4	0.00	00:01:04
HQ	RMB - FUJITI	2/27/2007 2:20:16AM	Yes	1CA22093	G729A	4	0.00	00:01:05
HQ	RMB - FUJITI	2/27/2007 3:05:39AM	Yes	6412110E	G729A	4	0.00	00:01:04
HQ	RMB - FUJITI	2/27/2007 5:05:30AM	Yes	C29346A7	G729A	3	0.00	00:01:04
HQ	RMB - FUJITI	2/27/2007 5:20:33AM	Yes	E7A99C5B	G729A	2	0.00	00:01:05
HQ	RMB - FUJITI	2/27/2007 5:35:34AM	Yes	6D46D03A	G729A	3	0.00	00:01:04

Figure C-15 Media Stream Detail Report

Field	Presence/Frequency	Description
Site A	Once for each media stream.	The name of the site. This data is retrieved from the ASiteName field in the MediaStream table.
Site B	Once for each media stream.	The name of the site that communicates to Site A. This data is retrieved from the BSiteName field in the MediaStream table.
Start Time	Once for each media stream.	The time the media stream began. This data is retrieved from the StartTime field in the MediaStream table.
WAN	Once for each media stream	“Yes” indicates the media stream accessed the WAN. “No” indicates the media stream did not access the WAN.
CallID	Once for each media stream.	The Call Identification number for the media stream listed on the detail report. By matching the CallID in the report to the CallID of a WAN call with voice quality issues, you can understand the cause of the problems. This data is retrieved from the CallID field in the MediaStream table.
Encoding	Once for each media stream.	The method of voice encoding used for the media stream. This data is retrieved from the EncodingType field in the MediaStream table. The EncodingType field can have the following values for encoding methods: 1=ALAW, 2=MULAW, 3=LINEAR, 4=ADPCM, 5=G729A, or 6=G729B.
Max Jitter	Once for each media stream.	The maximum jitter encountered. This value is the maximum of the A MaxJitter or B MaxJitter for the record in the MediaStream table for the media stream. If a significant number of calls are reported with a Max Jitter value close or equal to the Maximum Jitter Buffer value, you may want to increase the Maximum Jitter Buffer or investigate the cause of excess jitter in the network.
% Packet Loss	Once for each media stream.	This is the number of packets that were expected to arrive but did not arrive at the destination. Lost packets were mostly likely dropped on their way through the network.
Duration	Once for each media stream.	The duration of time the media stream was used across the WAN connection. This data is retrieve from the DurationSeconds field of the MediaStream table record for this call.

Table C-11 Media Stream Detail Report field descriptions

C.4.11 Account Code Summary Report

Summarizes call information for each account, including number of calls each day, along with their total and average duration. There are also totals for the reporting period. This report allows the administrator to indicate whether there should be summary information for each account. If this is desired, each extension's use of the account is summarized. If not, there's a simple total for the entire account code.

Account Codes are only applicable to outgoing calls. Outgoing calls are identified in the Call table of the CDR database with the CallType field set to 3 (Outbound). The only outgoing calls that appear in the report are those calls for which an Account Code was collected (the account code is recorded in the BillingCode field of the Call table). If the user does not provide an account code on an outgoing call (because it isn't required, or it is optional and they choose not to provide it), that call does not show up on the report.

Account Summary Report			
Start: 4/10/2003		End: 4/18/2003	
Start: 8:06:12AM		End: 11:06:12PM	
		Total Calls	Total Duration
			Average Duration
1234(test)			
	04/15/2003	4	00:01:55
	04/16/2003	6	00:01:40
	04/17/2003	5	00:06:53
	04/18/2003	2	00:01:57
	Total	17	00:12:25
1234(Test ACCT)			
	04/11/2003	1	00:03:57
	04/14/2003	6	00:01:24
	Total	7	00:05:21
Grand Total		24	00:17:46

Figure C-16 Account Code Summary Report

The Account Code Summary Report fields are described Table C-12.

C.4.12 Account Code Detail Report

Provides a detailed list of calls that occurred for each account. For each call the date and time of the call, number dialed, the extension making the call, and the duration of the call is included. For each account, a summary is provided of the number of calls, along with their total and average duration.

Account codes are only applicable to outgoing calls. Outgoing calls are identified in the Call table of the CDR database with the CallType field set to 3 (Outbound). The only outgoing calls that appear in the report are those calls for which an account code was collected (the account code is recorded in the BillingCode field of the Call table). If the user does not provide an account code on an outgoing call (because it isn't required, or it is optional and they choose not to provide it) that call does not show up on the report.

The Account Code Detail Report fields are described below.

C.5 CDR Database

This appendix specifies how data is stored in the CDR database tables. The CDR database records call data in the following tables:

Field	Presence/Frequency	Description
Account	Shown once for each account being reported on.	The account code that users enter to identify the account that a call is logged against. The name of the account code as configured in Director is also shown. For example, “300 (Marketing),” where “300” is the account code and “Marketing” is the name. For each call where account code information is collected, the account code is stored in the BillingCode field of the Call table. The name of the account code is stored in the FriendlyBillingCode field.
User	Only shown if the report option is selected to “Enable User Breakdown.” Shown once for each user who made calls for the account being reported upon.	The name and extension of the user who originated calls summarized in the report. For example, “John Smith (x3415).” The Extension field in the Call table identifies the party who originated the account code call. The extension field is where the report gets the extension number information. The name of the user comes from the PartyID (the first name) and PartyIDLastName (the last name) fields of the Connect record for the party that originated the call. The Connect record is tied to the Call table record, by the CallTableID field in the Connect table. All the Connect records for a particular call have the same CallTableID setting.
Total Calls	Repeated for each row.	The total number of calls for a particular day for the account. The total is broken down by user within each account if the “Enable User Breakdown” option is selected. A call is reported for the day on which the call started. That is, if a call starts on one day but ends on the next day, it is only reported for the day on which it started. The start of the call comes from the StartTime field in the Call table record for each call.
Total Duration	Repeated for each row.	The total duration of the calls being reported on the row. The duration is the total call duration, even if the call was transferred to parties such that the originator of the call was not on the call for the entire period. Duration is reported in the day the call started, but includes the entire call duration. For example, a call starts on 1/17 (with 20 minutes on 1/17) and ends on 1/18 (with 30 minutes on 1/18). The call is reported on 1/17 with duration of 50 minutes. This is then included in the total duration for all calls on 1/17. The duration of each call in this report comes from the Duration field in the call table.
Average Duration	Repeated for each row.	Calculated by dividing the total duration for a row by total calls.

Table C-12 Account Code Summary Report field descriptions

- **Call Table:** An entry is made in the Call table for each call in the ShoreTel 6.1 system. Other tables often reference the entries to the Call table.
- **Connect Table:** An entry is made in the Connect table for each connection to a call. When used with the Call table, a complete call history is provided.
- **MediaStream Table:** An entry is made in the MediaStream table each time there is a media stream between two switches that are at different sites. In some cases, such as for conference calls, there may be multiple media streams per call.
- **AgentActivity Table:** An entry is made in the AgentActivity table each time a workgroup agent logs into a workgroup and when he or she completes wrap-up.
- **QueueCall Table:** An entry is made in the QueueCall table for each call that is handled by a workgroup server. The entry identifies how the call leaves the workgroup—either by abandonment or for handling.
- **QueueStep Table:** An entry is made in the QueueStep table for each step where the workgroup server either hunts for agents or walks through workgroup queue steps. This provides more detailed information about how the call was disposed of by the workgroup server.
- **QueueDepth Table:** An entry is made in the QueueDepth table each time the depth of a workgroup server's call queue changes.

Account Detail Report					
Start: 4/10/2003				End: 4/10/2003	
Start: 8:07:32AM				End: 11:07:32PM	
Date	Time	Dialed Number	Calling Extension	Duration	
1234(test)					
4/15/2003	3:49:41PM	+14087671234	303	00:00:10	
4/15/2003	4:17:46PM	+14083313910	303	00:00:54	
4/15/2003	4:19:14PM	+14083313910	303	00:00:30	
4/15/2003	4:20:29PM	+14083313910	303	00:00:21	
4/17/2003	4:20:14PM	+14086159912	302	00:03:04	
4/18/2003	9:38:45AM	+14082183184	302	00:00:37	
4/18/2003	9:52:11AM	+18312471647	302	00:01:20	
4/16/2003	9:42:55AM	+14087671234	303	00:00:12	
4/16/2003	10:01:43AM	+14087671234	303	00:00:48	
4/16/2003	11:47:29AM	+14083313861	303	00:00:10	
4/16/2003	11:52:55AM	+14087671234	303	00:00:07	
4/16/2003	11:53:10AM	+14083313861	303	00:00:11	
4/16/2003	12:25:33PM	+14087671234	202	00:00:12	
4/17/2003	9:38:32AM	914087671212	301	00:00:35	
4/17/2003	9:39:34AM	+14087671234	301	00:00:16	
4/17/2003	9:43:25AM	914087671212	301	00:00:06	
4/17/2003	11:42:25AM	+14082183184	302	00:02:52	
Total		17 Call(s)	Total Duration: 00:12:25	Average Duration: 00:00:43	
1234(Test & CCT)					
4/11/2003	11:48:29AM	+14083065559	303	00:03:57	
4/14/2003	2:34:45PM	+14087671234	303	00:00:08	
4/14/2003	2:35:27PM	+14087671234	303	00:00:07	
4/14/2003	2:36:04PM	+14087671234	303	00:00:08	
4/14/2003	2:36:22PM	+14083313913	303	00:00:09	
4/14/2003	2:37:08PM	+14087671234	303	00:00:28	
4/14/2003	3:40:25PM	+14083313913	303	00:00:24	
Total		7 Call(s)	Total Duration: 00:05:21	Average Duration: 00:00:45	
Grand Total		24 Call(s)		00:17:46 Total	00:00:44 Average

Figure C-17 Account Code Detail Report

Field	Presence/Frequency	Description
Account	Shown once for each account being reported on	The account code that users enter to identify the account that a call is logged against. The name of the account code as configured in Director is also shown. For example, "300 (Marketing)," where "300" is the account code and "Marketing" is the name. For each call where account code information is collected, the account code is stored in the BillingCode field of the Call table. The name of the account code is stored in the FriendlyBillingCode field.
Date	Repeated for each call.	The date on which the call started. A call is reported for the day on which the call started. That is, if a call starts on one day, but ends on the next day, it is only reported for the day that it started on. The Date is extracted from the StartTime field in the Call record for each call in the report.
Time	Repeated for each call.	The time at which the call started. Time comes from the StartTime field in the Call table record for each call in the report.
Dialed Number	Repeated for each call.	The number that was dialed to begin the call. Dialed Number comes from the DialedNumber field in the Call table record for each call in the report.
Calling Extension	Repeated for each call.	The number of the user that originated the call. Calling Extension comes from the Extension field in the Call table record for each call in the report.
Duration	Repeated for each call.	The total duration of the calls being reported on the row. The duration is the call duration, even if the call was transferred to parties such that the originator of the call was not on the call for the entire period. The duration of each call in this report comes from the Duration field in the Call table.

Table C-13 Account Code Detail Report field descriptions

In addition to the tables listed above, the database contains a number of enumeration tables, which are documented below when discussing the tables that reference these enumeration/lookup tables.

Logged data is reflective of the time of the logging. For example, certain records contain the name of a trunk group from the configuration database. The name of the trunk group may be changed in the configuration database. New log entries reflect the changed name, but existing logs continue to have the old name.

C.5.1 Call Table

The CDR database reflects all calls within the system with a few exceptions which are listed below. These exceptions reflect the ShoreWare Telephony Management Server (TMS) that allows calls to continue even when portions of the system or network are not available. As the TAPI service provider for the ShoreWare Server, TMS manages the call control communications between all other ShoreWare services.

The exceptions are:

- If TMS is not connected to any of the call endpoints, the call is not recorded in the Call table. Because of network outages, TMS may not be connected to call endpoints, yet the call endpoints may have the connectivity necessary to complete the call (for example, the switches are able to communicate with each other but not to TMS).
- If TMS is not connected to some of the call endpoints (for example, a switch involved in the call), the information about the call can be incomplete (for example, the information in the Connect table as explained in the next section would only reflect some of the parties involved in the call).
- If TMS is restarted, any call entries that were incomplete, along with their associated Connect entries are destroyed. Incomplete calls do not show “Yes” in the locked field.
- Also at TMS restart time, TMS logs any calls in progress.

The following flow chart illustrates how new entries are added to the Call table whenever there is a call in the ShoreTel 6.1 system. Note that an entry is added to the Call table when the call begins (or when TMS starts up, for any calls in progress) and is updated when the call ends.

The Call table is reference by other tables, most important among them being the Connect table. You can analyze the Call and Connect tables to understand the complete disposition of a call as attempts are made to add parties, transfers occur, and so on. Other tables can index the Call table, through the primary key “ID,” which is unique for each record.

There is a CallID field that is used internally by the ShoreTel 6.1 system to identify calls. This, however, should not be used as the index into the table).

Close examination of the Call table shows that there are more calls recorded than you may initially expect. For example, if a call is made to a workgroup, you will see an initial call, generally from an incoming trunk. As agents are hunted, calls are made by the workgroup server to agents. If multiple agents are hunted, there will be multiple calls. Once one of the agents is successfully hunted, if you looked at the Connect table you see the agent being attached to the original call.

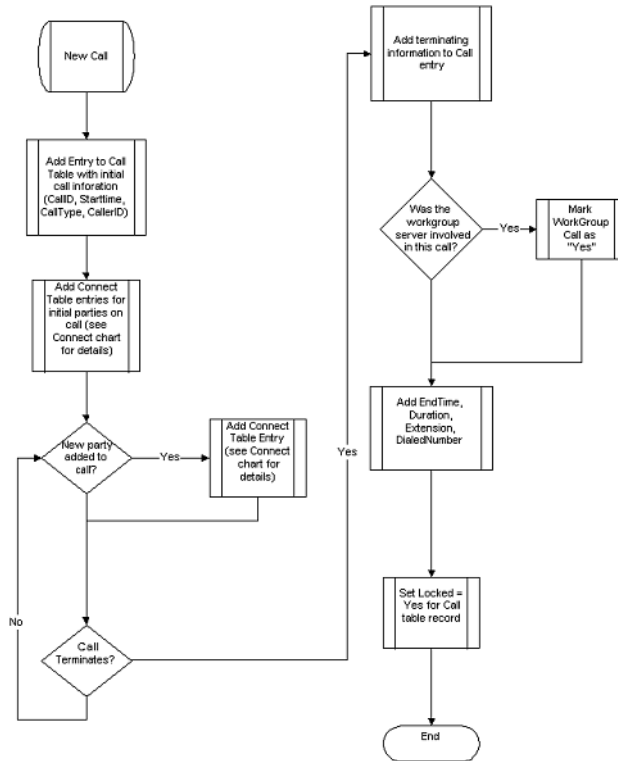


Figure C-18 New Entries in the Call Table

Field Name	Data Type	Description
ID	AutoNumber	Unique identifier. (4-byte integer, required)
CallID	Number	Unique number for the existence of the call. (4-byte integer)
SIP GUID	Text	SIP Unique Global ID number (32 characters, zero-length)
StartTime	Date/Time	For an inbound call, this is when the trunk has been seized. For an outbound call, this is when the user has completed dialing. (8-byte date/time, required)
StartTimeMS	Number	Append this information to the StartTime to reduce the absolute start time to the millisecond when the call began. (2-byte integer, required)
EndTime	Date/Time	Time when the call terminates (either by the near end hanging up or when the end external to the system hangs up) and the ShoreTel switch receives the notification of the disconnect. (8-byte date/time)
EndTimeMS	Number	Append this information to the StartTime to reduce the absolute start time to the milliseconds of when the call began (milliseconds). (2-byte integer, required)
CallNote	Text	User entered Call Note. This can be added from the ShoreTel desktop client. (64 characters, zero-length)
BillingCode	Text	Account code assigned to the call. (32 characters, zero-length)
Locked	Yes/No	Read-only status for this call (set once call has ended). Not locked means the call is still in progress. (boolean)

Table C-14 Call Table field descriptions

Field Name	Data Type	Description
Extension	Text	<p>For an outbound or extension-to-extension call, the extension has the dialed number of the originator of the call.</p> <p>This field is blank for an outbound call from an anonymous phone with no currently assigned DN.</p> <p>For an inbound call, the extension field contains the DN of the last party involved in the call (excluding voice mail or auto-attendant). For instance, an incoming call to an extension that transferred the call to extension 300 has "300" in the extension field (the complete history of parties connecting to the call can be found on the Connect table).</p> <p>All calls to an extension that forwarded to voice mail have the extension of the called party and not the voice mail number. (15 characters, zero-length)</p>
Duration	Date/Time	Elapsed time of the call from beginning to end. Calculated by subtracting StartTime from EndTime. Start time begins when the first party is added to a call. End time is when the last party leaves resulting in the end of the call. (8-byte date/time).
CallType	Number	See enumeration in CallType table. (1-byte integer, required)
WorkGroupCall	Yes/No	<p>Is this a workgroup call? Yes indicates that the workgroup server was involved in processing the call.</p> <p>If the call was directed toward a workgroup server, but that server was unavailable, then this field is set to "No" because the workgroup server never becomes involved in the call. (boolean)</p>
LongDistance	Yes/No	<p>From the perspective of the trunk for the call, did this call involve a long distance connection? The first connect record of the call is used to determine whether a call is long distance. If the first leg is an extension call, the value is always No.</p> <p>A trunk call can be transferred or conferenced, so the total long distance time can only be determined by examining all Connect records. (boolean)</p>
DialedNumber	Text	Extension-to-extension and outbound: Number dialed plus trunk access code if any. (15 characters, zero-length)
CallerID	Text	For CallType=Inbound only: Caller-ID number if present. If blocked or unavailable text is provided by the PSTN to indicate caller ID as unavailable it is included here; for example, the text may be "blocked" or "unavailable" (15 characters, zero-length)
Archived	Yes/No	Has this call been archived? (boolean)

Table C-14 Call Table field descriptions

C.5.2 Enumeration Tables: Use for the Call Table

C.5.2.1 Call Type

Call Type	Name	Description
0	Null	
1	ExtToExt	Extension-to-extension call.
2	Inbound	A trunk is the originating party.
3	Outbound	An extension is originating and a trunk is called.

Table C-15 Call Type descriptions

C.6 Connect Table

The Connect table contains a record for each party in a call. There are many different types of parties that can be reflected in the table including individual user extensions, workgroups, workgroup agents, and trunks.

The following flow chart illustrates how new entries are added to the Connect table each time a party is added to a call within the ShoreTel 6.1 system.

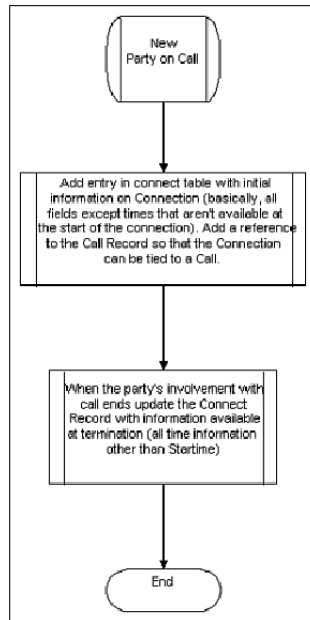


Figure C-19 New Entries in the Connect Table

C.6.1 Enumeration Tables Used for Connect Table

C.6.1.1 PartyType

Table C-16 lists the Connect Table party types.

Type #	Party Type	Description
0	Null	
1	Station	User extension which is currently assigned a home port; sometimes referred to as a "logged in user".
2	Trunk	Trunk (of any kind).
3	Virtual	A user extension which does not currently have an assigned home port (sometimes referred to as a "logged out user").
4	Workgroup	Workgroup extension.
5	AutoAttendant	Auto-Attendant extension.
6	VMForward	Voice mail forward extension (take a message).
7	VMLogin	Voice mail login extension.
8	BackupAA	Backup auto-attendant (built into switch).
9	AnonPhone	Anonymous telephone.
10	Nightbell	Nightbell extension.

Table C-16 Party Type Enumeration Table

Type #	Party Type	Description
11	Paging	Paging extension.
12	WorkgroupAgent	Records marked as WorkgroupAgent for calls transferred from a Workgroup to an Agent. Direct inbound calls to an agent are Station type.
13	Unknown	Unknown type.
14	RoutePoint	Route point.

Table C-16 Party Type Enumeration Table

C.6.1.2 PartyIDFlag

Table C-17 lists the Connect Table party ID flags.

Flag #	Party ID Flag Name	Description
0	Null	
1	Blocked	Blocked
2	OutOfArea	Out-Of-Area
4	Name	Name
8	Address	Address
12	NameAddress	Name & Address
16	Partial	Partial
32	Unknown	Unknown
64	Unavailable	Unavailable

Table C-17 Party ID Flag Enumeration Table

C.6.1.3 ConnectReason

Table C-18 lists the Connect Table connect reason codes.

Connect #	Connect Reason	Description
0	Null	
1	Direct	TMS was not available when the party connected to the call. Connection information is logged, but there is no more ConnectReason information.
2	ForwardBusy	The party was connected because the previous party's call handling mode was set to forward calls if the previous party was busy.
3	ForwardNoAnswer	The party was connected because previous party's call handling mode was set to forward calls if the previous party didn't answer.
4	ForwardAll	The party was connected because previous party's call handling mode was set to forward all calls.
5	Pickup	The call was connected because the called party answered the call.
6	Unpark	Unpark
7	Redirect	Redirect
8	Completion	Completion
9	Transfer	The call was connected after the call was transferred to the party.
10	Reminder	Reminder
11	Unknown	Unknown
12	Unavailable	Unavailable

Table C-18 Connect Reason Enumeration Table

Connect #	Connect Reason	Description
13	Intrude	Intrude
14	Parked	Parked
15	CampedOn	CampedOn
16	RouteRequest	RouteRequest
17	Called	The party was added to the call because it was the initial target of the call.
18	Forward	Forward
19	Originate	The party initiated this call.
20	Conference	The party was added to the call because the party was conferenced into the call.

Table C-18 Connect Reason Enumeration Table

C.6.1.4 Disconnect Reasons

Table C-19 lists the Connect Table disconnect reason codes.

Reason #	Disconnect Reason	Description
0	Null	
1	Normal	Normal termination
2	Unknown	Unknown reason
3	Reject Call	Call was rejected
4	Pickup Call	Call picked up by other destination
5	Forwarded Call	Call forwarded to another destination
6	Busy	Busy destination
7	NoAnswer	No answer by destination
8	BadAddress	Bad address
9	Unreachable	Destination cannot be reached
10	Congestion	Inadequate bandwidth
11	Incompatible	Destination is incompatible
12	Unavailable	Destination is unavailable
13	NoDialTone	No dial tone from the trunk
14	NumberChanged	Destination number changed
15	OutOfOrder	Destination out of order
16	TempFailure	Temporary failure
17	QoSUnavailable	QoS not available
18	Blocked	Destination blocked
19	DoNotDisturb	Do not disturb
20	Cancelled	Call cancelled
21	Unpark	Call unparked to different destination

Table C-19 Disconnect Reason Enumeration Table

C.7 MediaStream Table

The MediaStream table logs media information about InterSite Calls. At a high level, there is one such entry for each InterSite call. Information about both parties involved in the call is recorded.

Field Name	Data Type	Description
ID	AutoNumber	Unique identifier. (4-byte integer, required)
CallID	Number	Unique number for the existence of the call. (4-byte integer)
SIP GUID	Text	SIP Unique Global ID number (32 characters, zero-length)
EncodingType	Number	Encoding type used for media stream. (1-byte integer)
PayloadSize	Number	Media payload size in bytes for each media packet. (4-byte integer)
StartTime	Date/Time	Date and time the call started.
Duration	Date/Time	Elapsed time of call from begin to end. (8-byte date/time)
DurationSeconds	Number	Elapsed seconds time of call from begin to end. (4-byte integer)
FailureCode	Number	Error code. See MediaFailureCode table for enumeration. (1-byte integer)
A PartyType	Number	Party A's type enumeration. See the PartyType table. (1-byte integer)
A SiteID	Number	Party A's Site ID. (4-byte integer)
A SiteName	Text	Party A's Site Name. (50 characters, zero-length)
A LineID	Number	TAPI permanent line ID for party A. (4-byte integer)
A Name	Text	Call type name for party A. (50 characters, zero-length)
A_Extension	Text	Call extension number for party A (32 characters, zero-length)
A IP Address	Text	Local IP Address for party A. (15 characters, zero-length)
A TotalPackets	Number	Total packets received by party A. (4-byte integer)
A LostPackets	Number	Total packets lost by party A. (4-byte integer)
A MaxJitter	Number	Maximum jitter (ms) for party A. (4-byte integer)
A Underruns	Number	Number of receive underruns for party A. (4-byte integer)
A Overruns	Number	Number of receive underruns for party A. (4-byte integer)
B PartyType	Number	Party B's type enumeration. (1-byte integer)
B SiteID	Number	Party B's Site ID. (4-byte integer)
B SiteName	Text	Party B's Site Name. (50 characters, zero-length)
B LineID	Number	TAPI permanent line ID for party B. (4-byte integer)
B Name	Text	Call type name for party B. (50 characters, zero-length)
B_Extension	Text	Call extension number for party B (32 characters, zero-length)
B IP Address	Text	Local IP Address for party B. (15 characters, zero-length)
B TotalPackets	Number	Total packets received by party B. (4-byte integer)
B LostPackets	Number	Total packets lost by party B. (4-byte integer)
B MaxJitter	Number	Maximum jitter (ms) for party B. (4-byte integer)
B Underruns	Number	Number of receive underruns for party B. (4-byte integer)
B Overruns	Number	Number of receive overruns for party B. (4-byte integer)
InterSite	Yes/No	Indicates a logged call is InterSite. Only Intersite calls are logged.
Archived	Yes/No	Has this entry been archived? (boolean)

Table C-20 Media Stream Data Field descriptions

C.8 AgentActivity Table

The AgentActivity Table records information about workgroup agent's availability. Entries are made to record agents' Login/Logout from the workgroup and to reflect their time in Wrapup mode. The following flow chart illustrates how new entries are added to the AgentActivity table.

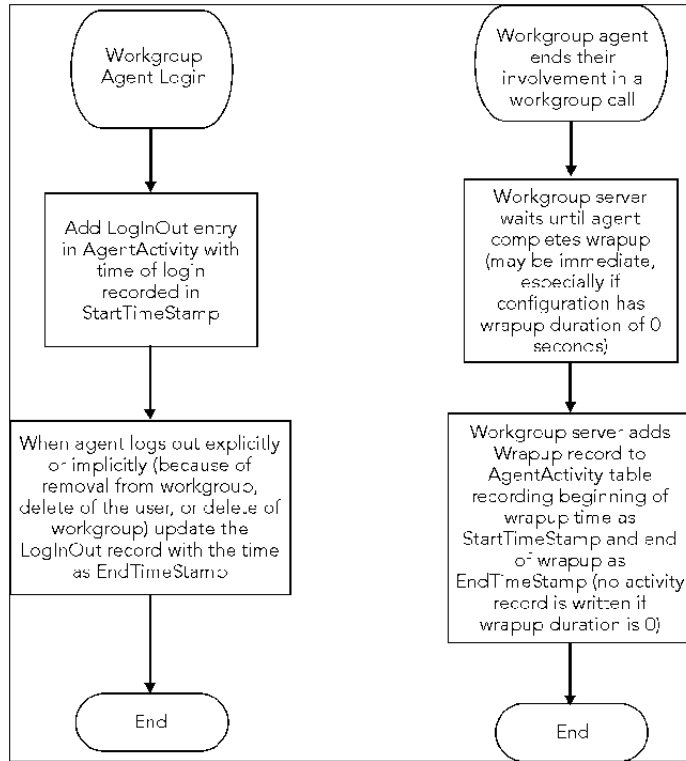


Figure C-20 Entries to the AgentActivity Table

The left flow shows how each time a workgroup agent logs in, a LoginOut entry is added, which is then updated at logout time. The right flow shows how the AgentActivity table is also updated as agents complete their handling of workgroup calls.

Field Name	Data Type	Description
ID	AutoNumber	Unique identifier. (4-byte integer, required)
CallID	Number	Unique number for the existence of the call. Provided in wrapup records. (4-byte integer)
AgentDN	Text	WorkGroup Agent's dialed number (extension). (15 characters, zero-length)
AgentFirstName	Text	WorkGroup Agent's First Name (50 Characters, zero-length).
AgentLastName	Text	WorkGroup Agent's Last Name (50 Characters, zero-length) (may be blank if the agent doesn't have a last name in the configuration database)
State	Number	Enumerated Agent State—set AgentStateLUT for possible values.
WorkGroupDN	Text	WorkGroup dialed number (extension) for which this agent activity is for (15 characters, zero-length)
WorkGroupName	Text	Workgroup's name. (50 Characters, zero-length)

Table C-21 Agent Activity Table field descriptions

Field Name	Data Type	Description
StartTimeStamp	Date/Time	Start time stamp. For LogInOut records, StartTimeStamp indicates the time when the agent logged into the workgroup. For wrapup records, the StartTimeStamp indicates the time when the agent entered wrapup time. See notes below. (8-byte date/time).
EndTimeStamp	Date/Time	End time stamp (8-byte date/time).
Archived	Yes/No	Has this entry been archived? (boolean)

Table C-21 Agent Activity Table field descriptions

- Two types of records are placed in the AgentActivity table. The State field identifies the type of record.
- LogInOut Records record the time that an agent is logged into the workgroup.
- Wrapup records record the time that an agent is in wrapup state.
- All records in the table should have ID, AgentDN, AgentFirstName, AgentLastName (unless blank), State, WorkGroupDN, WorkGroupName, StartTimeStamp, and Archived.
- LogInOut Records may exist for agents that have Logged into the workgroup but have not yet logged out. For these records the StartTimeStamp indicates the time when the agent logged into the workgroup. The EndTimeStamp is updated when the agent logs out of the workgroup with the time of the logout.
- For wrapup records the StartTimeStamp indicates the time when the agent entered wrapup time and EndTimeStamp indicates when they exit wrapup state.
- Wrapup records can contain a CallID to identify the Call that the agent was wrapping up from for the Wrapup record. This will not be provided in cases where the agent is manually placed in wrapup state when not on a call.
- There is always a wrapup record when an agent wraps up a call, even for the case where wrapup time is set to zero.

C.8.1 Enumeration Tables Used for AgentActivity

Table C-22 lists the Agent Activity Enumeration Tables.

State #	AgentState	Description
0	Null	
1	Reserved	Previously used for Login
2	Reserved	Previously used for Logout
3	Wrap_Up	Agent performing post-call wrap-up
4	Reserved	Temporarily used for Outcalls
5	LogInOut	Agent Login later updated with Logout time.
6	SecLogInOut	Secondary login activity for agents belonging to multiple Workgroup.

Table C-22 Agent Activity Enumeration Table

Login and Logout are no longer used.

C.9 QueueCall Table

Each time a call is processed by the workgroup server, an entry is made in the QueueCall table. A workgroup is a call queuing mechanism, thus the name “QueueCall” in the CDR database.

Each time a call is made to a workgroup when the workgroup server is operational, an entry is made in the QueueCall table; moreover, there is only one entry for each call. In other words, one and only one entry appears for each call. A call can be made to the workgroup dialed number, but if the workgroup server does not process the call, an entry is not made in the QueueCall table for the call. Moreover, the call will not be marked as a workgroup call in the call table.

The following flows chart illustrates how updates are made to the QueueCall table.

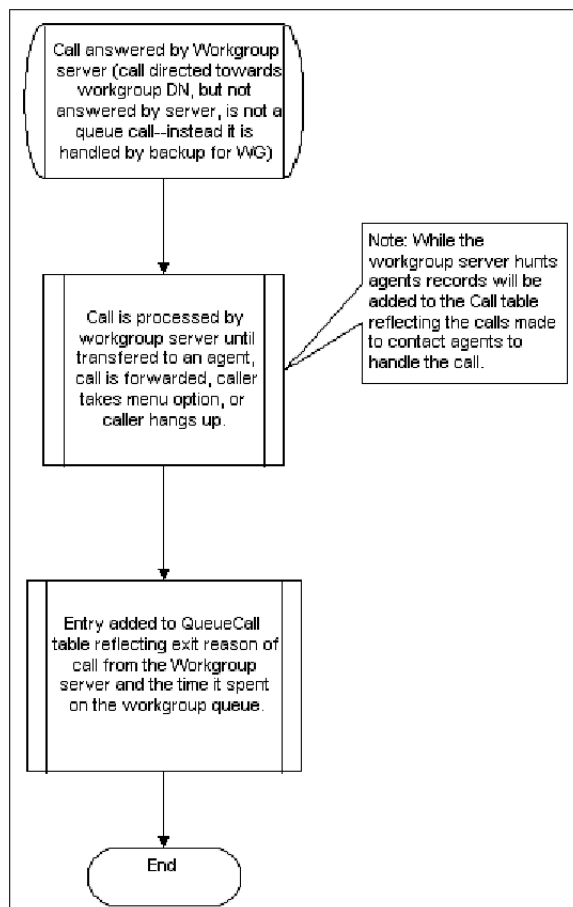


Figure C-21 New Entries to the CallQueue Table

Each entry in the QueueCall table contains the following fields:

- Partial records are never written. A record is written only once, either when the call is abandoned, the call is connected to an agent, or leaves the queue for other reasons as enumerated in QueueExitReasonLUT.
- If QueueExitReason = Abandon, target information (TargetType, TargetFirstName, TargetLastName, TargetDN) is meaningless and will be blank.
- If QueueExitReason is TransferToAgent, the TargetFirstName and TargetLastName are filled in with information about the agent.

Field Name	Data Type	Description
ID	AutoNumber	Unique identifier. (4-byte integer, required)
CallID	Number	Unique number for the existence of the call. (4-byte integer)
ConnectTableID	Number	Link to Connect Table ID Key. You can find more information about the connection to the call in the connect table. The Connect table entry here is for the Workgroup DNs connection to the call. If you want information from the Call table entry for this call, the reference to the Call table in the Connect entry should be used to find the Call table entry. (4-byte integer)
StartTime	Date/Time	The time at which the call is answered by the workgroup server, thereby beginning it's time on the call queue (workgroup) DN. (8-byte date/time)
Duration	Date/Time	Time from when the call is offered to the workgroup DN until it leaves the call queue. The call leaves the queue when it is answered by an agent, is abandoned by the calling party, or leaves the queue for other reasons. The complete lists of reasons for leaving the queue are found in the QueueExitReasonLUT table. (8-byte date/time)
DurationSeconds	Number	Duration expressed in number of seconds. (4-byte integer)
QueueName	Text	Name of the call queue (workgroup). (50 characters, zero-length)
QueueDN	Text	Extension number of the call queue (workgroup). (15 characters, zero-length)
ExitReason	Number	Enumerated reason the call left the call queue (see the QueueExitReasonLUT for enumerations). (1-byte integer)
TargetType	Number	Enumerated type of handoff target (see TargetTypeLUT for enumerations). (1-byte integer)
TargetFirstName	Text	Name or first name of target. (50 characters, zero-length)
TargetLastName	Text	Last name of target if applicable (blank if the target agent doesn't have a last name in the configuration database). (50 characters, zero-length)
TargetDN	Text	Dialed number of target. (15 characters, zero-length)
Archived	Yes/No	Has this entry been archived? (boolean)

Table C-23 Queue Call Table Field Descriptions

- If the QueueExitReason is Forwarding (2, 3, 4, or 5 for forward always, busy, no answer, or no logged in agent) or transfer (9, 10, and 11 for transfer to a menu, extension or voice mail), the DN that the call is being forwarded or transferred to is provided in the TargetDN field. However, the TargetFirstName and TargetLastName are not provided.
- A QueueExitReason is always entered. The field will never be blank. "Unknown" will only be used in the case of failure (and maybe not at all).

C.9.1 Enumeration Tables Used for QueueCall

ExitReason	Name	Description
0	Null	
1	TransferToAgent	Hunt succeeded and transferred to agent.
2	ForwardAlways	Workgroup forwarding all calls.
3	ForwardBusy	All logged in agents on call.
4	ForwardNoAnswer	All available agents did not answer.

Table C-24 Queue Call Exit Reason Enumeration Table

5	FwdNoLoginAgent	No logged in agents.
6	Reserved	
7	Abandon	Call dropped while in WG or Queue.
8	Reserved	
9	TransferVM	Option taken to transfer to voice mail
10	TransferExtension	Option taken to transfer to an extension.
11	TransferMenu	Option taken to transfer to a menu.
12	Pickup	Agent picked up call from queue.
13	Unpark	Agent parked call from queue.

Table C-24 Queue Call Exit Reason Enumeration Table

- ForwardMaxRings is no longer used.
- Exit Reasons for Forwarding (2-5) reflects the call being forwarded from the workgroup. These are used when the call leaves the workgroup as a result of call handling and the call handling indicates to forward the call to an internal or external number. Call handling can also indicate that the call is entering the call queue for the workgroup. In that case, these exit reasons are not used because the call does not exit the queue at that point.
- Exit Reason 8, Abandon, is used when the caller drops the call either by physically hanging up or by taking an option on a Queue Step to hang up.
- Even after a call is forwarded to the queue, it remains on the queue and it may still be successfully transferred to an agent or abandoned. Exit Reason 1 or 7 is recorded if either of these occurs.
- In addition to a call being successfully hunted or abandoned while on the queue, it may exit the queue because of an option taken during a queue step. The call will exit the queue if the caller takes any of the following options:
 - Take a message
 - Transfer to extension
 - Go to menu
 - Exit reasons 9, 10, and 11 have been added to cover these cases.

Target #	TargetType	Description
0	Null	
1	Agent	Workgroup agent.
2	Menu	A menu on the ShoreTel system.
3	Mailbox	A mailbox on the ShoreTel system.
4	OtherIntrnExtrn	Any other extensions to which the call is targeted.

Table C-25 Queue Call Target Type Enumeration Table

C.10 QueueStep Table

The QueueStep table logs data about time spent in queue steps or in hunting for agents.

There is a record for each period that the call spends hunting and for each period a call spends in a queue step. For example, if a call to a workgroup initially hunts for agents, then goes to the queue and exits the workgroup from that queue step, there will be two records

Field Name	Data Type	Description
ID	AutoNumber	Unique identifier. (4-byte integer, required)
QcallTableID	Number	Link to the Autonumber field in the QueueCall table. This essentially identifies the QueueCall that this step is associated with. (4-byte integer)
StartTime	Date/Time	Time at which the call first enters this step. (8-byte date/time)
Duration	Date/Time	Elapsed time spent in this step. (8-byte date/time)
DurationSeconds	Number	Elapsed seconds spent in this step. (4-byte integer)
StepNumber	Number	Step number if this is not a hunting record (as identified by the Hunting field set to Yes). The step number corresponds to the step number in the workgroup configuration.
ExitReason	Number	Enumerated reason for exit from step. (1-byte integer)
Hunting	Yes/No	If true the times correspond to hunting, or else this indicates a queue step. (boolean)

Table C-26 Queue Call Field Descriptions

for the call in the QueueStep table. The first record would be for hunting (the duration may be zero if, for example, no agents were logged in). The second record is for the first queue step from which the call exited.

ShoreGear Switches

This appendix describes the ShoreGear voice switches. Switch model numbers are located on the rear panel, as shown in Figure D-1.

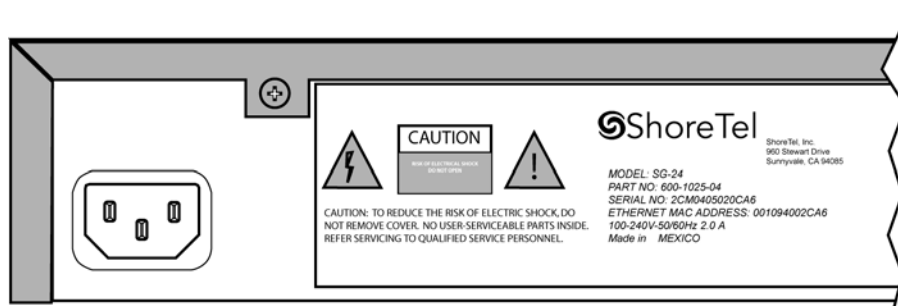


Figure D-1 Switch Model Number Label

D.1 Switch Models

ShoreGear models are classified into three switch families, based on chassis type:

- 1-U Half Width Switches
- 1-U Full Width Switches
- IPBX Switches

The following is a brief description of each switch family.

D.1.1 ShoreGear 1-U Half Width Voice Switches

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

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

ShoreGear 1-U Half Width Voice Switch models include:

- ShoreGear 30 (SG 30)
- ShoreGear 30BRI (SG 30BRI)
- ShoreGear 50 (SG 50)
- ShoreGear 90 (SG 90)

- ShoreGear 90BRI (SG 90BRI)
- ShoreGear 220T1 (SG 220T1)
- ShoreGear 220T1A (SG 220T1A)
- ShoreGear T1k (SG T1k)
- ShoreGear 220E1 (SG 220E1)
- ShoreGear E1k (SG E1k)

D.1.2 ShoreGear Voicemail Model Voice Switches

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

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

ShoreGear V Model Switch models include:

- ShoreGear 90V (SG 90V)
- ShoreGear 50V (SG 50V)
- ShoreGear 90BRIV (SG 90BRIV)

D.1.2.1 Capacity

Number of V Model switches allowed per system

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

Simultaneous Voicemail Calls per V Model switches

Voicemail Model Switches support the following number of simultaneous voicemail calls.

- SG50V – Maximum of 5 Voicemail calls per switch
 - G711 calls: 5
 - G729 calls: 2
- SG90V – Maximum of 9 Voicemail calls per switch
 - G711 calls: 9
 - G729 calls: 4
- SG90BRIV – Maximum of 9 Voicemail calls per switch
 - G711 calls: 9
 - G729 calls: 4

Call Load

Voicemail Model Switches call load capacity is as follows:

- 5400 BHCC when supporting 90 MGCP IP Phones or 90 SIP Trunks
- 3600 BHCC when supporting 90 SIP IP Phones or 90 SIP Trunks

Compact Flash Memory

Voicemail Model switches store voicemail and Auto Attendant files on compact flash. Flash card capacity for V Model Switches is:

- SG50V: 1 Gb
- SG90V: 2 Gb
- SG90BRIV: 2 Gb

Media Support

Voicemail Model Switches support the following media streams:

- G711
 - Music on Hold (MOH): 15 calls
 - Backup Auto Attendant (BAA): 50 calls
- G729
 - Music on Hold (MOH): none
 - Backup Auto Attendant (BAA): none

SIP support

Voicemail Model Switches support the following SIP media streams:

- G711 Ringback tone (Hunt Groups and Work Group calls): 50 media streams
- G729 Ringback tone (Hunt Groups and Work Group calls): no support

D.1.3 ShoreGear 1-U Full Width Voice Switches

The ShoreGear 1-U Full Width Switch family includes three models that support analog, IP, SIP, T1, and E1 voice data streams. Full width switch models can be stacked or mounted directly into a standard 19-inch equipment rack. These switches are all 1 RU and have an RJ21X connector for connection to analog phones and trunks. They also feature redundant Ethernet LAN connections for greater availability and reliability.

ShoreGear 1-U Full Width Voice Switch models include:

- ShoreGear 120 (SG 120) – also referred to as ShoreGear 120/24 (SG 120/24)
- ShoreGear 60 (SG 60) – also referred to as ShoreGear 60/12 (SG 60/12)
- ShoreGear 40 (SG 48) – also referred to as ShoreGear 40/8 (SG 40/8)
- ShoreGear T1 (SG T1)
- ShoreGear E1 (SG E1)
- ShoreGear 24A (SG 24A)

D.1.4 ShoreGear IPBX Voice Switches

The ShoreGear IPBX Switch family is the earliest ShoreGear switch design. Although ShoreTel no longer markets IPBX switches, existing switches are supported and can be used in a ShoreTel network configuration. IPBX switches are full width and varying heights, depending on the switch model.

ShoreGear IPBX Voice Switch models include:

- ShoreGear 24 (IPBX 24) Voice Switch (2 RU)
- ShoreGear T1 (IPBX T1) Voice Switch (1.5 RU)
- ShoreGear E1 (IPBX E1) Voice Switch (1.5 RU)

D.2 Specifications – SG 1-U Half-Width Switches

D.2.1 ShoreGear 90 Voice Switch

The following sections describe ShoreGear 90 resource capacity, LED behavior, and connectors. The ShoreGear 90 is not supported in installations outside the U.S. and Canada. Figure D-2 displays the ShoreGear 90 front plate.

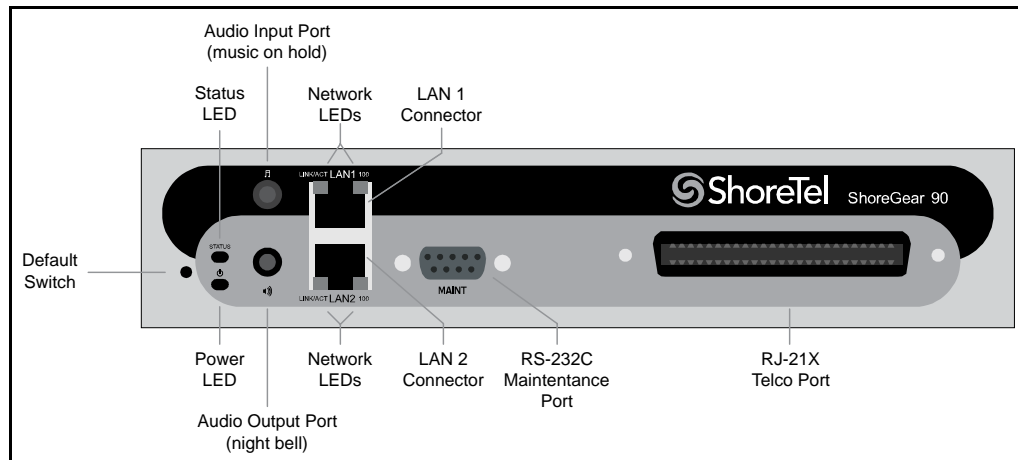


Figure D-2 ShoreGear 90 Front Plate

D.2.1.1 Switch Capacity

- Analog Circuit Resources
 - Ports 1-8: Eight Loop Start Trunks
 - Ports 9-12: Four Extensions or DID Trunks. A single command configures all ports as either Extensions or DID trunks.
 - Power Failure Transfer Unit: Trunk Port 1 to Extension Port 12
- Make Me Conference Resources: 12 ports
 - Ports 1-12
- Maximum IP Phone Resources: 90 devices
 - Analog Port Reallocation: 60
 - Built-in Resources: 30

D.2.1.2 LED Descriptions

Power LED

The ShoreGear 90 has one power LED, which indicates the following:

- On: The switch is operating normally.
- Off: The switch has no power.
- Flashing:

- **2 flashes**—The switch failed its internal self-test. This indicates a hardware failure. Replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
- **3 flashes**—Booting via FTP. Flash memory might be corrupted. Go to the **Quick Look** page to ensure that the system is running properly.
- **4 flashes**—The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.
- **5 flashes**—The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreWare server, and set the boot file to `/tsk/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreWare server's IP address, and set option 67 to `/tsk/vxworks`.
- **6 flashes**—Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

The ShoreGear 90 network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:
 - *When flashing*, network activity is detected.
 - *When on (not flashing)*, the switch is connected to an Ethernet network.
 - *When off*, the switch cannot detect an Ethernet network.

This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- **100M:**
 - *When green*, the switch is connected to a 100BaseT network.
 - *When off*, the switch is connected to a 10BaseT network.

Status LED

The ShoreGear 90 has one status LED to provide general information about the ports. The color and blink pattern of the LED indicate the port function:

- **Status LED (Green)**
 - *When on steady*, no ports are handling active calls.
 - *When flashing fast*, at least one port is handling an active call.
- **Status LED (Yellow)**
 - *When on steady*, no ports are handling active calls and at least one port is out of service.
 - *When flashing slow*, the switch is not connected (or has lost connection) to a ShoreTel server.
 - *When flashing fast*, at least one port is handling an active call and at least one port is out of service.
- **Off**: No ports are assigned.

D.2.1.3 ShoreGear 90 Connectors

The ShoreGear 90 voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-21X male connector for mass termination of the telephone/trunk ports
 - *Power Failure Transfer Unit*: Trunk Port 1 to Extension Port 12
 - *Backup Operator*: Extension Port 12

ShoreGear 90 RJ-21X Telephone and Trunk Connector

Table D-1 lists the RJ-21X Ring and Tip pin numbers for the SG 90

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
1	Trunk	1	Blue/White	26	White/Blue
–		2	Orange/White	27	White/Orange
2	Trunk	3	Green/White	28	White/Green
–		4	Brown/White	29	White/Brown
3	Trunk	5	Slate/White	30	White/Slate
–		6	Blue/Red	31	Red/Blue
4	Trunk	7	Orange/Red	32	Red/Orange
–		8	Green/Red	33	Red/Green
5	Trunk	9	Brown/Red	34	Red/Brown
–		10	Slate/Red	35	Red/Slate
6	Trunk	11	Blue/Black	36	Black/Blue
–		12	Orange/Black	37	Black/Orange
7	Trunk	13	Green/Black	38	Black/Green
–		14	Brown/Black	39	Black/Brown
8	Trunk	15	Slate/Black	40	Black/Slate
–		16	Blue/Yellow	41	Yellow/Blue
9	Extension - DID	17	Orange/Yellow	42	Yellow/Orange

Table D-1 ShoreGear 90 RJ-21X Telephone and Trunk Connector Pins

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
–		18	Green/Yellow	43	Yellow/Green
10	Extension - DID	19	Brown/Yellow	44	Yellow/Brown
–		20	Slate/Yellow	45	Yellow/Slate
11	Extension - DID	21	Blue/Violet	46	Violet/Blue
–		22	Orange/Violet	47	Violet/Orange
12	Extension - DID	23	Green/Violet	48	Violet/Green
–		24	Brown/Violet	49	Violet/Brown
–		25	Slate/Violet	50	Violet/Slate

Table D-1 ShoreGear 90 RJ-21X Telephone and Trunk Connector Pins

D.2.2 ShoreGear 90BRI Voice Switch

The following sections describe ShoreGear 90BRI resource capacity, LED behavior, and connectors. Figure D-3 displays the ShoreGear 90BRI front plate.

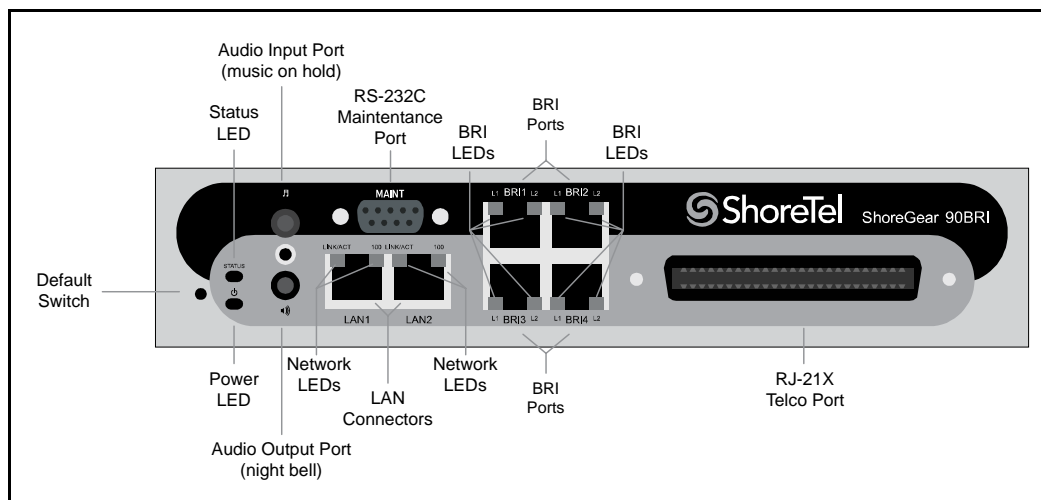


Figure D-3 ShoreGear 90BRI Front Plate

D.2.2.1 Switch Capacity

- Analog Circuit Resources
 - Ports 9-12: Extensions
- Digital Circuit Resources
 - Four BRI Spans, each comprising two channels: Eight channels maximum
- Make Me Conference Resource: None
- Maximum IP Phone Resources: 90 devices
 - Analog Port Reallocation: 20
 - Digital Channel Reallocation: 40
 - Built-in Resources: 30

D.2.2.2 LED Descriptions

Power LED

The ShoreGear 90BRI has one power LED, which indicates the following:

- **On:** The switch is operating normally.
- **Off:** The switch has no power.
- **Flashing**
 - **2 flashes**—The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
 - **3 flashes**—Booting via FTP. Flash memory might be corrupted. Go to the **Quick Look** page to ensure that the system is running properly.
 - **4 flashes**—The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.
 - **5 flashes**—The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreWare server, and set the boot file to `/tsk/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreWare server's IP address, and set option 67 to `/tsk/vxworks`.
 - **6 flashes**—Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

The ShoreGear 90BRI network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:
 - When flashing, network activity is detected.
 - When on (not flashing), the switch is connected to an Ethernet network.
 - When off, the switch cannot detect an Ethernet network.

This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- 100M
 - *When green*, the switch is connected to a 100BaseT network.
 - *When off*, the switch is connected to a 10BaseT network.

Status LED

The ShoreGear 90BRI has one status LED to provide general information about the ports. The color and blink pattern of the LED indicate the port function:

- Status LED (Green)
 - *When on steady*, no ports are handling active calls.
 - *When flashing fast* (100 msec on/off), at least one port is handling an active call.
- Status LED (Yellow)
 - *When on steady*, no ports are handling active calls and at least one port is out of service.
 - *When flashing slow* (1 sec. on/off), the switch is not connected (or has lost connection) to a ShoreTel server.
 - *When flashing fast* (100 msec on/off), at least one port is handling an active call and at least one port is out of service.
- Off: No ports are assigned.

BRI LED

Each BRI connector has two LEDs to indicate port activity. The color and blink pattern of the LED indicate the port function:

- LED 1: Off, LED 2 Off – Port not configured in Director
- LED 1: Yellow, LED 2 Off – Port inactive or not connected
- LED 1: Off, LED 2 Off – Layer 1 active. Layer 2 not established
- LED 1: Off, LED 2 Green – Layer 1 active. Layer 2 active.
- LED 1: Off, LED 2 Green flashing – Call in progress (Layer 1, Layer 2, and Layer 3 active).

D.2.2.3 ShoreGear 90BRI Connectors

The ShoreGear 90BRI voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-21X male connector for mass termination of the telephone/trunk ports
- 4 RJ-45 T1 telco port

ShoreGear 90BRI RJ-21X Telephone and Trunk Connector

Table D-2 lists the RJ-21X Ring and Tip pin numbers for the SG 90BRI

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
–		1	Blue/White	26	White/Blue
–		2	Orange/White	27	White/Orange
–		3	Green/White	28	White/Green
–		4	Brown/White	29	White/Brown
–		5	Slate/White	30	White/Slate
–		6	Blue/Red	31	Red/Blue
–		7	Orange/Red	32	Red/Orange
–		8	Green/Red	33	Red/Green
–		9	Brown/Red	34	Red/Brown
–		10	Slate/Red	35	Red/Slate
–		11	Blue/Black	36	Black/Blue
–		12	Orange/Black	37	Black/Orange
–		13	Green/Black	38	Black/Green
–		14	Brown/Black	39	Black/Brown
–		15	Slate/Black	40	Black/Slate
–		16	Blue/Yellow	41	Yellow/Blue
9	Extension	17	Orange/Yellow	42	Yellow/Orange
–		18	Green/Yellow	43	Yellow/Green
10	Extension	19	Brown/Yellow	44	Yellow/Brown
–		20	Slate/Yellow	45	Yellow/Slate
11	Extension	21	Blue/Violet	46	Violet/Blue
–		22	Orange/Violet	47	Violet/Orange
12	Extension	23	Green/Violet	48	Violet/Green
–		24	Brown/Violet	49	Violet/Brown
–		25	Slate/Violet	50	Violet/Slate

Table D-2 ShoreGear 90BRI RJ-21X Telephone and Trunk Connector Pins

D.2.3 ShoreGear 50 Voice Switch

The following sections describe ShoreGear 50 resource capacity, LED behavior, and connectors. The ShoreGear 50 is not supported in installations outside the U.S. and Canada. Figure D-4 displays the ShoreGear 50 front plate.

D.2.3.1 Switch Capacity

- Analog Circuit Resources
 - Ports 1-4: Four Loop Start Trunks
 - Ports 11-12: Two Extensions or DID Trunks. A single command configures all ports as either Extensions or DID trunks.
 - Power Failure Transfer Unit: Trunk Port 1 to Extension Port 12
- Make Me Conference Resources: six ports
 - Ports 1-4, 11-12

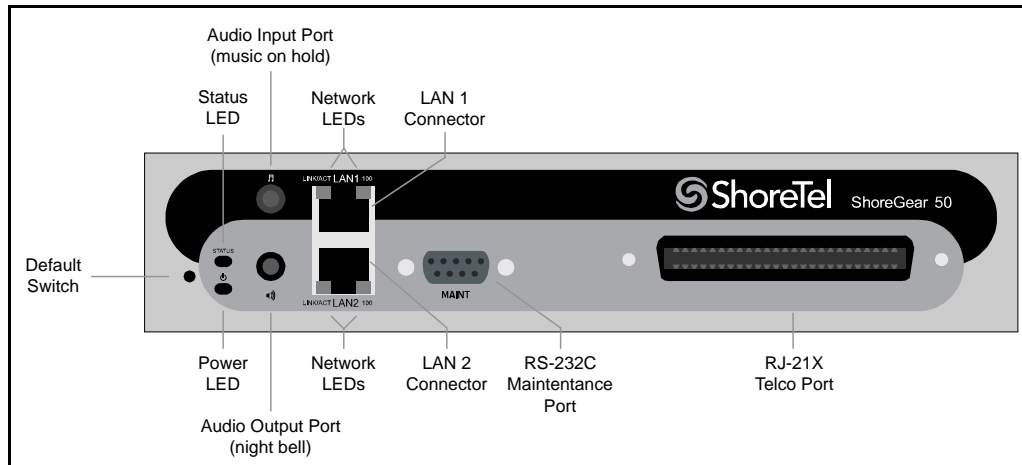


Figure D-4 ShoreGear 50 Front Plate

- Maximum IP Phone Resources: 50 devices
 - Analog Port Reallocation: 30
 - Built-in Resources: 20

D.2.3.2 LED Descriptions

Power LED

The ShoreGear 50 has one power LED, which indicates the following:

- On: The switch is operating normally.
- Off: The switch has no power.
- Flashing
 - *2 flashes*—The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
 - *3 flashes*—Booting via FTP. Flash memory might be corrupted. Go to the **Quick Look** page to ensure that the system is running properly.
 - *4 flashes*—The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.
 - *5 flashes*—The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreWare server, and set the boot file to `/tsk/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreWare server's IP address, and set option 67 to `/tsk/vxworks`.
 - *6 flashes*—Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If

the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

The ShoreGear 50 network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:

- *When flashing*, network activity is detected.
- *When on (not flashing)*, the switch is connected to an Ethernet network.
- *When off*, the switch cannot detect an Ethernet network.

This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- **100M**
 - *When green*, the switch is connected to a 100BaseT network.
 - *When off*, the switch is connected to a 10BaseT network.

Status LED

The ShoreGear 50 has one status LED to provide general information about the ports. The color and blink pattern of the LED indicate the port function:

- **Status LED (Green)**
 - *When on steady*, no ports are handling active calls.
 - *When flashing fast*, at least one port is handling an active call.
- **Status LED (Yellow)**
 - *When on steady*, no ports are handling active calls and at least one port is out of service.
 - *When flashing slow*, the switch is not connected (or has lost connection) to a ShoreTel server.
 - *When flashing fast*, at least one port is handling an active call and at least one port is out of service.
- **Off:** No ports are assigned.

D.2.3.3 ShoreGear 50 Connectors

The ShoreGear 50 voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface

- 1 RJ-21X male connector for mass termination of the telephone/trunk ports
 - *Power Failure Transfer Unit*: Trunk Port 1 to Extension Port 12
 - *Backup Operator*: Extension Port 12

ShoreGear 50 RJ-21X Telephone and Trunk Connector

Table D-3 lists the RJ-21X Ring and Tip pin numbers for the SG 50

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
1	Trunk	1	Blue/White	26	White/Blue
–		2	Orange/White	27	White/Orange
2	Trunk	3	Green/White	28	White/Green
–		4	Brown/White	29	White/Brown
3	Trunk	5	Slate/White	30	White/Slate
–		6	Blue/Red	31	Red/Blue
4	Trunk	7	Orange/Red	32	Red/Orange
–		8	Green/Red	33	Red/Green
–		9	Brown/Red	34	Red/Brown
–		10	Slate/Red	35	Red/Slate
–		11	Blue/Black	36	Black/Blue
–		12	Orange/Black	37	Black/Orange
–		13	Green/Black	38	Black/Green
–		14	Brown/Black	39	Black/Brown
–		15	Slate/Black	40	Black/Slate
–		16	Blue/Yellow	41	Yellow/Blue
9	Extension - DID	17	Orange/Yellow	42	Yellow/Orange
–		18	Green/Yellow	43	Yellow/Green
10	Extension - DID	19	Brown/Yellow	44	Yellow/Brown
–		20	Slate/Yellow	45	Yellow/Slate
11	Extension - DID	21	Blue/Violet	46	Violet/Blue
–		22	Orange/Violet	47	Violet/Orange
12	Extension - DID	23	Green/Violet	48	Violet/Green
–		24	Brown/Violet	49	Violet/Brown
–		25	Slate/Violet	50	Violet/Slate

Table D-3 ShoreGear 50 RJ-21X Telephone and Trunk Connector Pins

D.2.4 ShoreGear 30 Voice Switch

The following sections describe ShoreGear 30 resource capacity, LED behavior, and connectors. The ShoreGear 30 is not supported in installations outside the U.S. and Canada. Figure D-5 displays the ShoreGear 30 front plate.

D.2.4.1 Switch Capacity

- Analog Circuit Resources

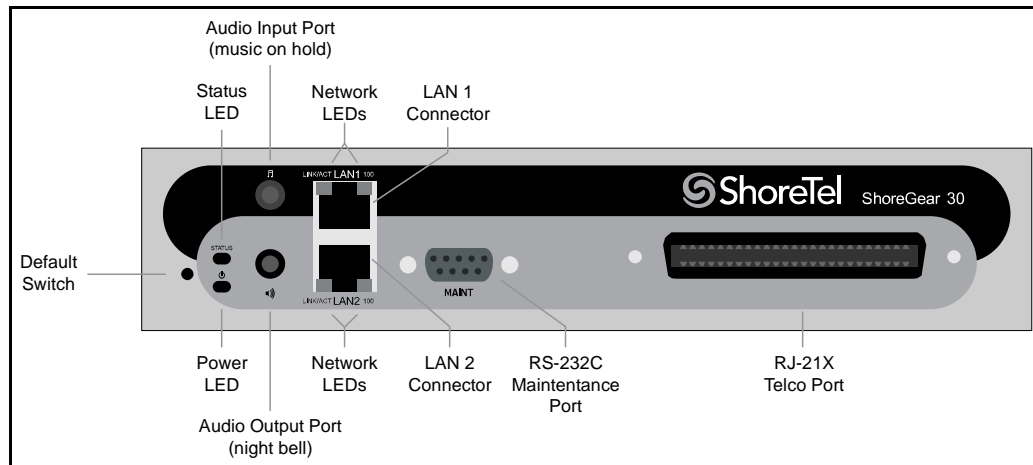


Figure D-5 ShoreGear 30 Front Plate

- *Ports 1-2*: Two Loop Start Trunks
- *Ports 11-12*: Two Extensions or DID Trunks. A single command configures all ports as either Extensions or DID trunks.
- *Power Failure Transfer Unit*: Trunk Port 1 to Extension Port 12
- **Make Me Conference Resources**: none
- **Maximum IP Phone Resources**: none
 - *Analog Port Reallocation*: 20
 - *Built-in Resources*: 10

D.2.4.2 LED Descriptions

Power LED

The ShoreGear 30 has one power LED, which indicates the following:

- **On**: The switch is operating normally.
- **Off**: The switch has no power.
- **Flashing**:
 - *2 flashes*—The switch failed its internal self-test. This indicates a hardware failure. Replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
 - *3 flashes*—Booting via FTP. Flash memory might be corrupted. Go to the **Quick Look** page to ensure that the system is running properly.
 - *4 flashes*—The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.
 - *5 flashes*—The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreWare server, and set the boot file to

/tsk/vxworks. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreWare server's IP address, and set option 67 to /tsk/vxworks.

- **6 flashes**—Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

The ShoreGear 30 network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:
 - *When flashing*, network activity is detected.
 - *When on (not flashing)*, the switch is connected to an Ethernet network.
 - *When off*, the switch cannot detect an Ethernet network.

This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- **100M:**
 - *When green*, the switch is connected to a 100BaseT network.
 - *When off*, the switch is connected to a 10BaseT network.

Status LED

The ShoreGear 30 has one status LED to provide general information about the ports. The color and blink pattern of the LED indicate the port function:

- **Status LED (Green)**
 - *When on steady*, no ports are handling active calls.
 - *When flashing fast*, at least one port is handling an active call.
- **Status LED (Yellow)**
 - *When on steady*, no ports are handling active calls and at least one port is out of service.
 - *When flashing slow*, the switch is not connected (or has lost connection) to a ShoreTel server.
 - *When flashing fast*, at least one port is handling an active call and at least one port is out of service.
- **Off:** No ports are assigned.

D.2.4.3 ShoreGear 30 Connectors

The ShoreGear 30 voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-21X male connector for mass termination of the telephone/trunk ports
 - *Power Failure Transfer Unit*: Trunk Port 1 to Extension Port 12
 - *Backup Operator*: Extension Port 12

ShoreGear 30 RJ-21X Telephone and Trunk Connector

Table D-4 lists the RJ-21X Ring and Tip pin numbers for the SG 30

Port	Type	Ring		Tip		
		Pin #	Cable Color	Pin #	Cable Color	
1	Trunk	1	Blue/White	26	White/Blue	
–		2	Orange/White	27	White/Orange	
2	Trunk	3	Green/White	28	White/Green	
–		4	Brown/White	29	White/Brown	
–		5	Slate/White	30	White/Slate	
–		6	Blue/Red	31	Red/Blue	
–		7	Orange/Red	32	Red/Orange	
–		8	Green/Red	33	Red/Green	
–		9	Brown/Red	34	Red/Brown	
–		10	Slate/Red	35	Red/Slate	
–		11	Blue/Black	36	Black/Blue	
–		12	Orange/Black	37	Black/Orange	
–		13	Green/Black	38	Black/Green	
–		14	Brown/Black	39	Black/Brown	
–		15	Slate/Black	40	Black/Slate	
–		16	Blue/Yellow	41	Yellow/Blue	
–		17	Orange/Yellow	42	Yellow/Orange	
–		18	Green/Yellow	43	Yellow/Green	
–		19	Brown/Yellow	44	Yellow/Brown	
–		20	Slate/Yellow	45	Yellow/Slate	
11		Extension - DID	21	Blue/Violet	46	Violet/Blue
–			22	Orange/Violet	47	Violet/Orange
12	Extension - DID	23	Green/Violet	48	Violet/Green	
–		24	Brown/Violet	49	Violet/Brown	
–		25	Slate/Violet	50	Violet/Slate	

Table D-4 ShoreGear 30 RJ-21X Telephone and Trunk Connector Pins

D.2.5 ShoreGear 30BRI Voice Switch

The following sections describe ShoreGear 30BRI resource capacity, LED behavior, and connectors. Figure D-6 displays the ShoreGear 30BRI front plate.

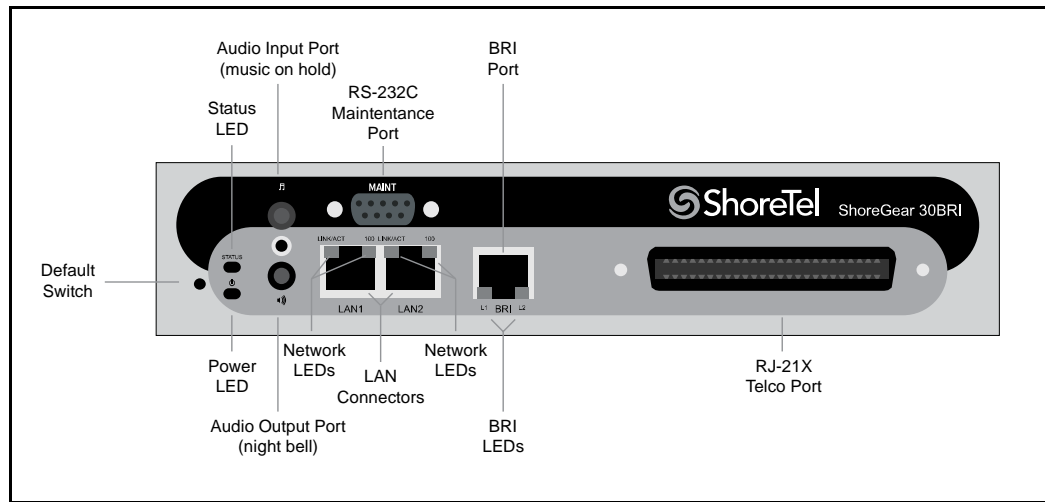


Figure D-6 ShoreGear 30BRI Front Plate

D.2.5.1 Switch Capacity

- Analog Circuit Resources
 - *Ports 11-12: Extensions*
- Digital Circuit Resources
 - *One BRI Span comprising two channels: two channels maximum*
- Make Me Conference Resource: None
- Maximum IP Phone Resources: 30 devices
 - *Analog Port Reallocation: 10*
 - *Digital Channel Reallocation: 10*
 - *Built-in Resources: 10*

D.2.5.2 LED Descriptions

Power LED

The ShoreGear 30BRI has one power LED, which indicates the following:

- On: The switch is operating normally.
- Off: The switch has no power.
- Flashing
 - *2 flashes*—The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
 - *3 flashes*—Booting via FTP. Flash memory might be corrupted. Go to the **Quick Look** page to ensure that the system is running properly.

- **4 flashes**—The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.
- **5 flashes**—The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreWare server, and set the boot file to `/tsk/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreWare server's IP address, and set option 67 to `/tsk/vxworks`.
- **6 flashes**—Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

The ShoreGear 30BRI network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:
 - When flashing, network activity is detected.
 - When on (not flashing), the switch is connected to an Ethernet network.
 - When off, the switch cannot detect an Ethernet network.

This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- **100M**
 - *When green*, the switch is connected to a 100BaseT network.
 - *When off*, the switch is connected to a 10BaseT network.

Status LED

The ShoreGear 30BRI has one status LED to provide general information about the ports. The color and blink pattern of the LED indicate the port function:

- **Status LED (Green)**
 - *When on steady*, no ports are handling active calls.
 - *When flashing fast* (100 msec on/off), at least one port is handling an active call.
- **Status LED (Yellow)**

- *When on steady*, no ports are handling active calls and at least one port is out of service.
 - *When flashing slow* (1 sec. on/off), the switch is not connected (or has lost connection) to a ShoreTel server.
 - *When flashing fast* (100 msec on/off), at least one port is handling an active call and at least one port is out of service.
- Off: No ports are assigned.

BRI LED

Each BRI connector has two LEDs to indicate port activity. The color and blink pattern of the LED indicate the port function:

- LED 1: Off, LED 2 Off – Port not configured in Director
- LED 1: Yellow, LED 2 Off – Port inactive or not connected
- LED 1: Off, LED 2 Off – Layer 1 active. Layer 2 not established
- LED 1: Off, LED 2 Green – Layer 1 active. Layer 2 active.
- LED 1: Off, LED 2 Green flashing – Call in progress (Layer 1, Layer 2, and Layer 3 active).

D.2.5.3 ShoreGear 30BRI Connectors

The ShoreGear 30BRI voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-21X male connector for mass termination of the telephone/trunk ports
- 4 RJ-45 T1 telco port

ShoreGear 30BRI RJ-21X Telephone and Trunk Connector

Table D-5 lists the RJ-21X Ring and Tip pin numbers for the SG 30BRI

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
–		1	Blue/White	26	White/Blue
–		2	Orange/White	27	White/Orange
–		3	Green/White	28	White/Green
–		4	Brown/White	29	White/Brown
–		5	Slate/White	30	White/Slate
–		6	Blue/Red	31	Red/Blue
–		7	Orange/Red	32	Red/Orange
–		8	Green/Red	33	Red/Green
–		9	Brown/Red	34	Red/Brown
–		10	Slate/Red	35	Red/Slate
–		11	Blue/Black	36	Black/Blue
–		12	Orange/Black	37	Black/Orange
–		13	Green/Black	38	Black/Green

Table D-5 ShoreGear 30BRI RJ-21X Telephone and Trunk Connector Pins

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
–		14	Brown/Black	39	Black/Brown
–		15	Slate/Black	40	Black/Slate
–		16	Blue/Yellow	41	Yellow/Blue
9	Extension	17	Orange/Yellow	42	Yellow/Orange
–		18	Green/Yellow	43	Yellow/Green
10	Extension	19	Brown/Yellow	44	Yellow/Brown
–		20	Slate/Yellow	45	Yellow/Slate
11	Extension	21	Blue/Violet	46	Violet/Blue
–		22	Orange/Violet	47	Violet/Orange
12	Extension	23	Green/Violet	48	Violet/Green
–		24	Brown/Violet	49	Violet/Brown
–		25	Slate/Violet	50	Violet/Slate

Table D-5 ShoreGear 30BRI RJ-21X Telephone and Trunk Connector Pins

D.2.6 ShoreGear 220T1 Voice Switch

The following sections describe ShoreGear 220T1 resource capacity, LED behavior, and connectors. The ShoreGear 220T1 is not supported in installations outside the U.S. and Canada. Figure D-7 displays the ShoreGear 220T1 front plate.

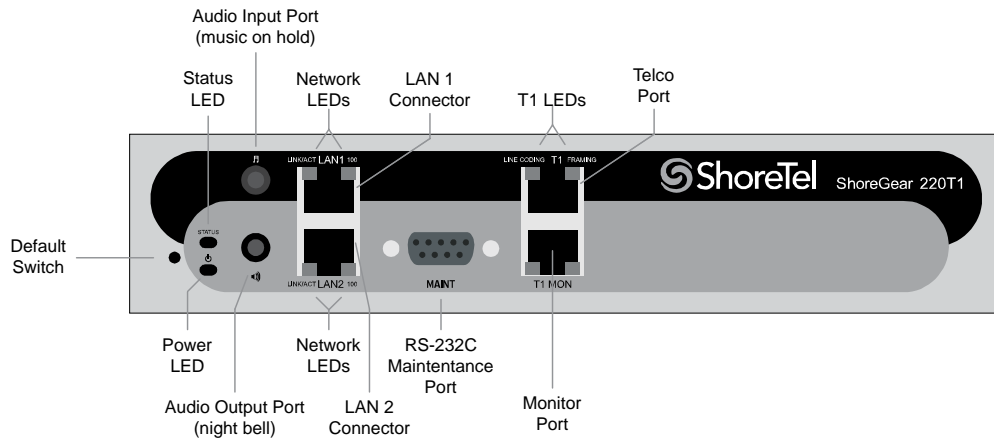


Figure D-7 ShoreGear 220T1 Front Plate

D.2.6.1 Switch Capacity

- Digital Circuit Resources: 24 channels maximum
 - One T1 circuit, 24 channels per circuit: 24 channels maximum
- Make Me Conference Resource: None
- Maximum IP Phone Resources: 220
 - Digital Channel Reallocation: 120
 - Built-in Resources: 100

D.2.6.2 LED Descriptions

Power LED

The ShoreGear 220T1 has one power LED, which indicates the following:

- **On:** The switch is operating normally.
- **Off:** The switch has no power.
- **Flashing**
 - **2 flashes**—The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
 - **3 flashes**—Booting via FTP. Flash memory might be corrupted. Go to the **Quick Look** page to ensure that the system is running properly.
 - **4 flashes**—The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.
 - **5 flashes**—The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreWare server, and set the boot file to `/tsk/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreWare server's IP address, and set option 67 to `/tsk/vxworks`.
 - **6 flashes**—Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

The ShoreGear 220T1 network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:
 - **When flashing**, network activity is detected.
 - **When on (not flashing)**, the switch is connected to an Ethernet network.
 - **When off**, the switch cannot detect an Ethernet network.

This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- 100M
 - *When green*, the switch is connected to a 100BaseT network.
 - *When off*, the switch is connected to a 10BaseT network.

Status LED

The ShoreGear 220T1 has one status LED to provide general information about the ports. The color and blink pattern of the LED indicate the port function:

- Status LED (Green)
 - *When on steady*, no ports are handling active calls.
 - *When flashing fast*, at least one port is handling an active call.
- Status LED (Yellow)
 - *When on steady*, no ports are handling active calls and at least one port is out of service.
 - *When flashing slow*, the switch is not connected (or has lost connection) to a ShoreTel server.
 - *When flashing fast*, at least one port is handling an active call and at least one port is out of service.
- Off: No ports are assigned.

Monitor and Telco LEDs

The Monitor and Telco LEDs indicate line coding, network framing, and loopback status. These LEDs are color coded—green, yellow, and red. The Monitor and Telco LED descriptions follow.

Telco and Monitor LED alarms and errors are logged as switch events in ShoreWare Director's event log.

- **Line Coding:** This LED indicates line coding status, as follows:
 - *When green*, the line coding signal is good.
 - *When yellow*, bipolar violations (BPV) are being received at one second intervals.
 - *When red*, a loss of signal (LOS) has occurred.
 - *When flashing red*, loopback is active (local or CO).
 - *When off*, the switch has no power.
- **Framing:** This LED indicates network framing status, as follows:
 - *When green*, the T1/E1 signal is *in frame*; the signal is synchronized.
 - *When yellow*, a *yellow alarm* has been received from the Central Office.
 - *When flashing yellow*, the frame-bit error rate has exceeded its limits. A small number of frame-bit errors (>1 per million) have occurred; this state will take up to 10 minutes to clear.
 - *When flashing fast yellow*, a series of frame-bit errors (>1 per 1000) have occurred.
 - *When red*, the T1/E1 signal is out-of-frame (OOF). The received signal cannot be framed to the Extended Superframe (ESF) or D4 format.
 - *When flashing red*, loopback is active (local or CO).
 - *When off*, the switch has no power.

D.2.6.3 ShoreGear 220T1 Connectors

The ShoreGear 220T1 voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-45 T1 telco port
- 1 RJ-45 T1 monitor port for connecting test equipment

D.2.7 ShoreGear 220T1A Voice Switch

The following sections describe ShoreGear 220T1A resource capacity, LED behavior, and connectors. The ShoreGear 220T1A is not supported in installations outside the U.S. and Canada. Figure D-8 displays the ShoreGear 220T1A front plate.

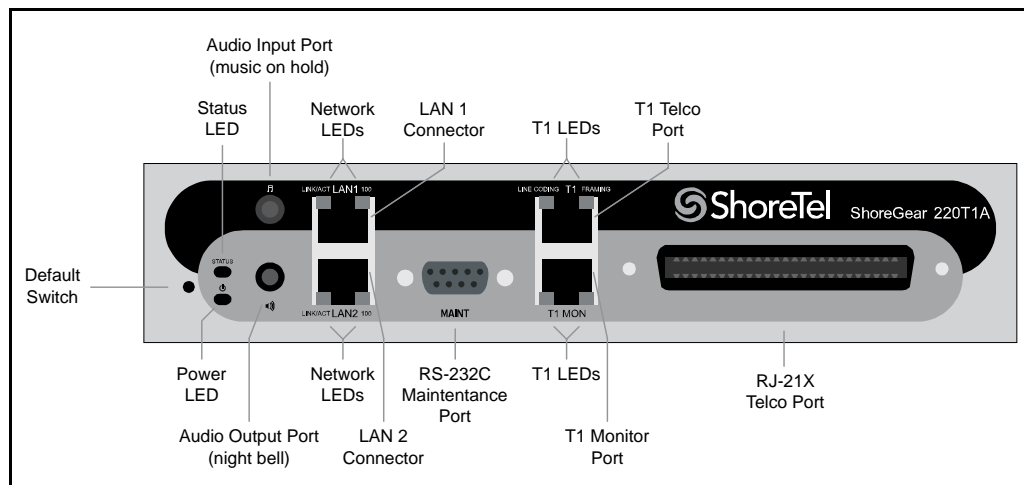


Figure D-8 ShoreGear 220T1A Front Plate

D.2.7.1 Switch Capacity

- **Analog Circuit Resources**
 - *Ports 1-2:* Two Loop Start Trunks
 - *Ports 9-12:* Four Extensions or DID Trunks. A single command configures all ports as either Extensions or DID trunks.
 - *Power Failure Transfer Unit:* Trunk Port 1 to Extension Port 12
- **Digital Circuit Resources:** 24 channels maximum
 - *One T1 circuit, 24 channels per circuit*
- **Make Me Conference Resource:** Six ports
 - *Ports 1-2, 9-12*
- **Maximum IP Phone Resources:** 220 devices
 - *Analog Channel Reallocation:* 30
 - *Digital Channel Reallocation:* 120
 - *Built-in Resources:* 70

D.2.7.2 LED Descriptions

Power LED

The ShoreGear 220T1A has one power LED, which indicates the following:

- **On:** The switch is operating normally.
- **Off:** The switch has no power.
- **Flashing**
 - **2 flashes**—The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
 - **3 flashes**—Booting via FTP. Flash memory might be corrupted. Go to the **Quick Look** page to ensure that the system is running properly.
 - **4 flashes**—The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.
 - **5 flashes**—The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreWare server, and set the boot file to `/tsk/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreWare server's IP address, and set option 67 to `/tsk/vxworks`.
 - **6 flashes**—Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

The ShoreGear 220T1A network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:
 - **When flashing**, network activity is detected.
 - **When on (not flashing)**, the switch is connected to an Ethernet network.
 - **When off**, the switch cannot detect an Ethernet network.

This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- 100M
 - *When green*, the switch is connected to a 100BaseT network.
 - *When off*, the switch is connected to a 10BaseT network.

Status LED

The ShoreGear 220T1A has one status LED to provide general information about the ports. The color and blink pattern of the LED indicate the port function:

- Status LED (Green)
 - *When on steady*, no ports are handling active calls.
 - *When flashing fast* (100 msec on/off), at least one port is handling an active call.
- Status LED (Yellow)
 - *When on steady*, no ports are handling active calls and at least one port is out of service.
 - *When flashing slow* (1 sec. on/off), the switch is not connected (or has lost connection) to a ShoreTel server.
 - *When flashing fast* (100 msec on/off), at least one port is handling an active call and at least one port is out of service.
- Off: No ports are assigned.

Monitor and Telco LEDs

The Monitor and Telco LEDs indicate line coding, network framing, and loopback status. These LEDs are color coded—green, yellow, and red. The Monitor and Telco LED descriptions follow.

Telco and Monitor LED alarms and errors are logged as switch events in ShoreWare Director's event log.

- **Line Coding:** This LED indicates line coding status, as follows:
 - *When green*, the line coding signal is good.
 - *When yellow*, bipolar violations (BPV) are received at one second intervals.
 - *When red*, a loss of signal (LOS) has occurred.
 - *When flashing red*, loopback is active (local or CO).
 - *When off*, the switch has no power.
- **Framing:** This LED indicates network framing status, as follows:
 - *When green*, the T1/E1 signal is *in frame*; the signal is synchronized.
 - *When yellow*, a *yellow alarm* has been received from the Central Office.
 - *When flashing yellow*, the frame-bit error rate has exceeded its limits.
 - *When flashing slow yellow*, a small number of frame-bit errors (>1 per million) have occurred; this state will take up to 10 minutes to clear.
 - *When flashing fast yellow*, a series of frame-bit errors (> 1 per 1000) have occurred.
 - *When red*, the T1/E1 signal is out-of-frame (OOF). The received signal cannot be framed to the Extended Superframe (ESF) or D4 format.
 - *When flashing red*, loopback is active (local or CO).
 - *When off*, the switch has no power.

D.2.7.3 ShoreGear 220T1A Connectors

The ShoreGear 220T1A voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 1 RJ-21X male connector for mass termination of the telephone/trunk ports
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-45 T1 telco port
- 1 RJ-45 T1 monitor port for connecting test equipment

ShoreGear 220T1A RJ-21X Telephone and Trunk Connector

Table D-6 lists the RJ-21X Ring and Tip pin numbers for the SG 220T1AI

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
1	Trunk	1	Blue/White	26	White/Blue
–		2	Orange/White	27	White/Orange
2	Trunk	3	Green/White	28	White/Green
–		4	Brown/White	29	White/Brown
–		5	Slate/White	30	White/Slate
–		6	Blue/Red	31	Red/Blue
–		7	Orange/Red	32	Red/Orange
–		8	Green/Red	33	Red/Green
–		9	Brown/Red	34	Red/Brown
–		10	Slate/Red	35	Red/Slate
–		11	Blue/Black	36	Black/Blue
–		12	Orange/Black	37	Black/Orange
–		13	Green/Black	38	Black/Green
–		14	Brown/Black	39	Black/Brown
–		15	Slate/Black	40	Black/Slate
–		16	Blue/Yellow	41	Yellow/Blue
9	Extension	17	Orange/Yellow	42	Yellow/Orange
–		18	Green/Yellow	43	Yellow/Green
10	Extension	19	Brown/Yellow	44	Yellow/Brown
–		20	Slate/Yellow	45	Yellow/Slate
11	Extension	21	Blue/Violet	46	Violet/Blue
–		22	Orange/Violet	47	Violet/Orange
12	Extension	23	Green/Violet	48	Violet/Green
–		24	Brown/Violet	49	Violet/Brown
–		25	Slate/Violet	50	Violet/Slate

Table D-6 ShoreGear 220T1A RJ-21X Telephone and Trunk Connector Pins

D.2.8 ShoreGear-220E1 Voice Switch

The following sections describe ShoreGear-220E1 resource capacity, LED behavior, and connectors. Figure D-9 displays the ShoreGear-220E1 front plate.

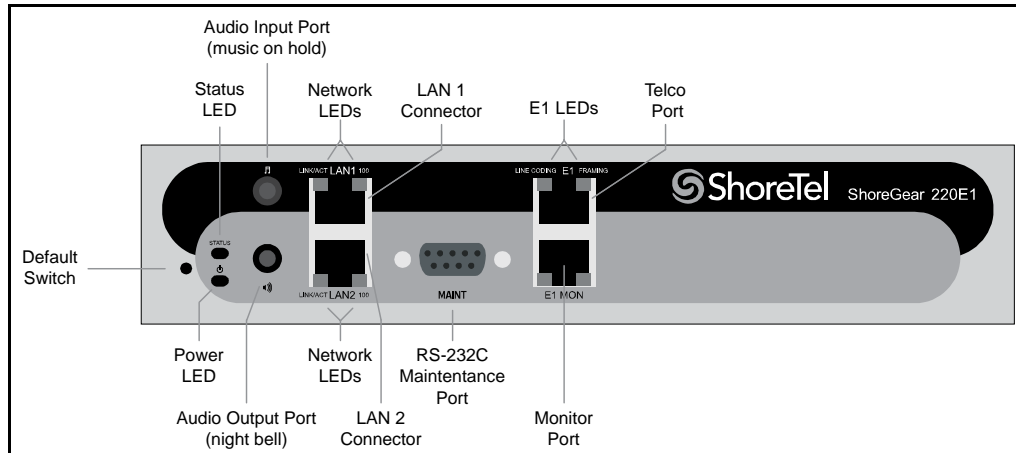


Figure D-9 ShoreGear-220E1 Front Plate

D.2.8.1 Switch Capacity

- Digital Circuit Resources: 30 channels maximum
 - One E1 circuit, 30 channels per circuit
- Make Me Conference Resource: none
- Maximum IP Phone Resources: 220
 - Digital Channel Reallocation: 150
 - Built-in Resources: 70

D.2.8.2 LED Descriptions

Power LED

The ShoreGear-220E1 has one power LED, which indicates the following:

- On: The switch is operating normally.
- Off: The switch has no power.
- Flashing
 - 2 *flashes*—The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
 - 3 *flashes*—Booting via FTP. Flash memory might be corrupted. Go to the **Quick Look** page to ensure that the system is running properly.
 - 4 *flashes*—The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.
 - 5 *flashes*—The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreWare server, and set the boot file to `/tsk/vxworks`. If you are using a DHCP server that supports options 66 and 67,

set option 66 to the ShoreWare server's IP address, and set option 67 to `/tsk/vxworks`.

- **6 flashes**—Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

The ShoreGear-220E1 network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:
 - *When flashing*, network activity is detected.
 - *When on* (not flashing), the switch is connected to an Ethernet network.
 - *When off*, the switch cannot detect an Ethernet network.

This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- **100M**
 - *When green*, the switch is connected to a 100BaseT network.
 - *When off*, the switch is connected to a 10BaseT network.

Status LED

The ShoreGear-220E1 has one status LED to provide general information about the ports. The color and blink pattern of the LED indicate the port function:

- **Status LED (Green)**
 - *When on steady*, no ports are handling active calls.
 - *When flashing fast*, at least one port is handling an active call.
- **Status LED (Yellow)**
 - *When on steady*, no ports are handling active calls and at least one port is out of service.
 - *When flashing slow*, the switch is not connected (or has lost connection) to a ShoreTel server.
 - *When flashing fast*, at least one port is handling an active call and at least one port is out of service.
- **Off:** No ports are assigned.

Monitor and Telco LEDs

The Monitor and Telco LEDs indicate line coding, network framing, and loopback status. These LEDs are color coded—green, yellow, and red. The Monitor and Telco LED descriptions follow.

Telco and Monitor LED alarms and errors are logged as switch events in ShoreWare Director's event log.

- **Line Coding:** This LED indicates line coding status, as follows:
 - *When green*, the line coding signal is good.
 - *When yellow*, bipolar violations (BPV) are being received at one second intervals.
 - *When red*, a loss of signal (LOS) has occurred.
 - *When flashing red*, loopback is active (local or CO).
 - *When off*, the switch has no power.
- **Framing:** This LED indicates network framing status, as follows:
 - *When green*, the T1/E1 signal is *in frame*; the signal is synchronized.
 - *When yellow*, a *yellow alarm* has been received from the Central Office.
 - *When flashing yellow*, the frame-bit error rate has exceeded its limits.
 - *When flashing slow yellow*, a small number of frame-bit errors ($10e^{-6}$) have occurred; this state will take up to 10 minutes to clear.
 - *When flashing fast yellow*, a series of frame-bit errors ($10e^{-3}$) have occurred.
 - *When red*, the T1/E1 signal is out-of-frame (OOF). The received signal cannot be framed to the Extended Superframe (ESF) or D4 format.
 - *When flashing red*, loopback is active (local or CO).
 - *When off*, the switch has no power.

D.2.8.3 ShoreGear-220E1 Connectors

The ShoreGear-220E1 voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-45 T1 telco port
- 1 RJ-45 T1 monitor port for connecting test equipment

D.2.9 ShoreGear T1k Voice Switch

The following sections describe ShoreGear T1k resource capacity, LED behavior, and connectors. The ShoreGear T1k is not supported in installations outside the U.S. and Canada. Figure D-10 displays the ShoreGear T1k front plate.

D.2.9.1 Switch Capacity

- **Digital Circuit Resources:** 24 channels maximum
 - *One T1 circuit, 24 channels per circuit*
- **Make Me Conference Resource:** None
- **Maximum IP Phone Resources:** None

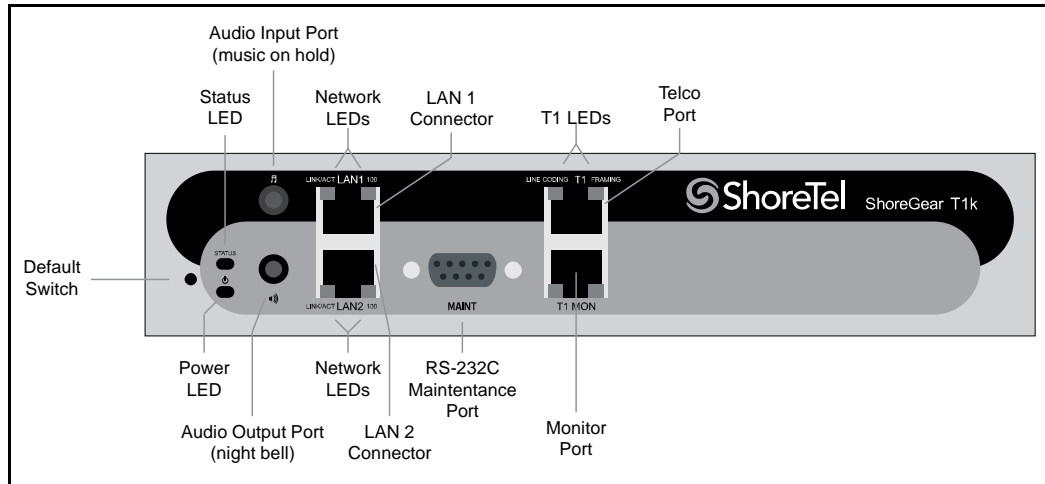


Figure D-10 ShoreGear T1k Front Plate

D.2.9.2 LED Descriptions

Power LED

The ShoreGear T1k has one power LED, which indicates the following:

- **On:** The switch is operating normally.
- **Off:** The switch has no power.
- **Flashing**
 - **2 flashes**—The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
 - **3 flashes**—Booting via FTP. Flash memory might be corrupted. Go to the **Quick Look** page to ensure that the system is running properly.
 - **4 flashes**—The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.
 - **5 flashes**—The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreWare server, and set the boot file to `/tsk/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreWare server's IP address, and set option 67 to `/tsk/vxworks`.
 - **6 flashes**—Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops

flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

The ShoreGear T1k network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:
 - *When flashing*, network activity is detected.
 - *When on (not flashing)*, the switch is connected to an Ethernet network.
 - *When off*, the switch cannot detect an Ethernet network.

This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- **100M**
 - *When green*, the switch is connected to a 100BaseT network.
 - *When off*, the switch is connected to a 10BaseT network.

Status LED

The ShoreGear T1k has one status LED to provide general information about the ports. The color and blink pattern of the LED indicate the port function:

- **Status LED (Green)**
 - *When on steady*, no ports are handling active calls.
 - *When flashing fast*, at least one port is handling an active call.
- **Status LED (Yellow)**
 - *When on steady*, no ports are handling active calls and at least one port is out of service.
 - *When flashing slow*, the switch is not connected (or has lost connection) to a ShoreTel server.
 - *When flashing fast*, at least one port is handling an active call and at least one port is out of service.
- **Off:** No ports are assigned.

Monitor and Telco LEDs

The Monitor and Telco LEDs indicate line coding, network framing, and loopback status. These LEDs are color coded—green, yellow, and red. The Monitor and Telco LED descriptions follow.

Telco and Monitor LED alarms and errors are logged as switch events in ShoreWare Director's event log.

- **Line Coding:** This LED indicates line coding status, as follows:
 - *When green*, the line coding signal is good.
 - *When yellow*, bipolar violations (BPV) are being received at one second intervals.

- When red, a loss of signal (LOS) has occurred.
- When flashing red, loopback is active (local or CO).
- When off, the switch has no power.
- Framing: This LED indicates network framing status, as follows:
 - When green, the T1/E1 signal is in frame; the signal is synchronized.
 - When yellow, a yellow alarm has been received from the Central Office.
 - When flashing yellow, the frame-bit error rate has exceeded its limits. A small number of frame-bit errors (>1 per million) have occurred; this state will take up to 10 minutes to clear.
 - When flashing fast yellow, a series of frame-bit errors (>1 per 1000) have occurred.
 - When red, the T1/E1 signal is out-of-frame (OOF). The received signal cannot be framed to the Extended Superframe (ESF) or D4 format.
 - When flashing red, loopback is active (local or CO).
 - When off, the switch has no power.

D.2.9.3 ShoreGear T1k Connectors

The ShoreGear T1k voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-45 T1 telco port
- 1 RJ-45 T1 monitor port for connecting test equipment

D.2.10 ShoreGear E1k Voice Switch

The following sections describe ShoreGear E1k resource capacity, LED behavior, and connectors. The ShoreGear E1k is not supported in installations outside the U.S. and Canada. Figure D-10 displays the ShoreGear E1k front plate.

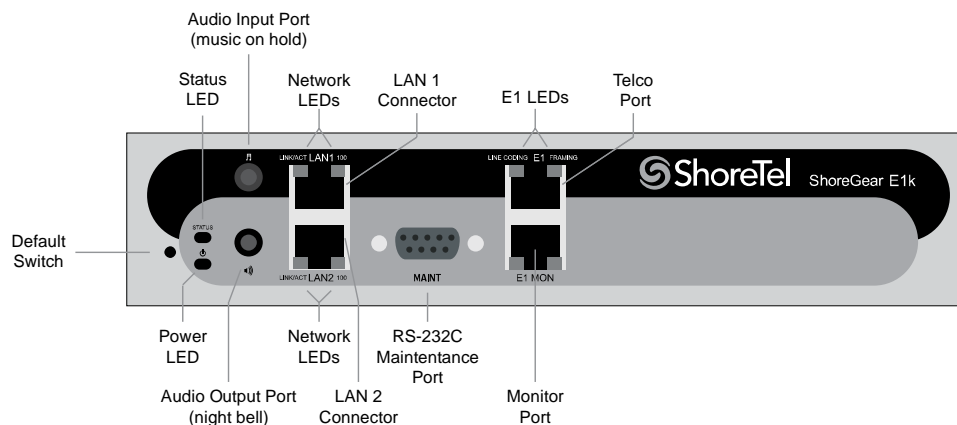


Figure D-11 ShoreGear E1k Front Plate

D.2.10.1 Switch Capacity

- Digital Circuit Resources: 30 channels maximum
 - One E1 circuit, 30 channels per circuit
- Make Me Conference Resource: None

- Maximum IP Phone Resources: None

D.2.10.2 LED Descriptions

Power LED

The ShoreGear E1k has one power LED, which indicates the following:

- **On:** The switch is operating normally.
- **Off:** The switch has no power.
- **Flashing**
 - **2 flashes**—The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
 - **3 flashes**—Bootting via FTP. Flash memory might be corrupted. Go to the **Quick Look** page to ensure that the system is running properly.
 - **4 flashes**—The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.
 - **5 flashes**—The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreWare server, and set the boot file to `/tsk/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreWare server's IP address, and set option 67 to `/tsk/vxworks`.
 - **6 flashes**—Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

The ShoreGear E1k network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:
 - *When flashing*, network activity is detected.
 - *When on (not flashing)*, the switch is connected to an Ethernet network.
 - *When off*, the switch cannot detect an Ethernet network.

This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- 100M
 - *When green*, the switch is connected to a 100BaseT network.
 - *When off*, the switch is connected to a 10BaseT network.

Status LED

The ShoreGear E1k has one status LED to provide general information about the ports. The color and blink pattern of the LED indicate the port function:

- Status LED (Green)
 - *When on steady*, no ports are handling active calls.
 - *When flashing fast*, at least one port is handling an active call.
- Status LED (Yellow)
 - *When on steady*, no ports are handling active calls and at least one port is out of service.
 - *When flashing slow*, the switch is not connected (or has lost connection) to a ShoreTel server.
 - *When flashing fast*, at least one port is handling an active call and at least one port is out of service.
- Off: No ports are assigned.

Monitor and Telco LEDs

The Monitor and Telco LEDs indicate line coding, network framing, and loopback status. These LEDs are color coded—green, yellow, and red. The Monitor and Telco LED descriptions follow.

Telco and Monitor LED alarms and errors are logged as switch events in ShoreWare Director's event log.

- **Line Coding:** This LED indicates line coding status, as follows:
 - *When green*, the line coding signal is good.
 - *When yellow*, bipolar violations (BPV) are being received at one second intervals.
 - *When red*, a loss of signal (LOS) has occurred.
 - *When flashing red*, loopback is active (local or CO).
 - *When off*, the switch has no power.
- **Framing:** This LED indicates network framing status, as follows:
 - *When green*, the E1 signal is *in frame*; the signal is synchronized.
 - *When yellow*, a *yellow alarm* has been received from the Central Office.
 - *When flashing yellow*, the frame-bit error rate has exceeded its limits. A small number of frame-bit errors (>1 per million) have occurred; this state will take up to 10 minutes to clear.
 - *When flashing fast yellow*, a series of frame-bit errors (>1 per 1000) have occurred.
 - *When red*, the T1/E1 signal is out-of-frame (OOF). The received signal cannot be framed to the Extended Superframe (ESF) or D4 format.
 - *When flashing red*, loopback is active (local or CO).
 - *When off*, the switch has no power.

D.2.10.3 ShoreGear E1k Connectors

The ShoreGear E1k voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-45 T1 telco port
- 1 RJ-45 T1 monitor port for connecting test equipment

D.3 Specifications – SG Voice Model Switches

D.3.1 ShoreGear 90V Voice Switch

The following sections describe ShoreGear 90V resource capacity, LED behavior, and connectors. The ShoreGear 90V is not supported in installations outside the U.S. and Canada. Figure D-12 displays the ShoreGear 90V front plate.

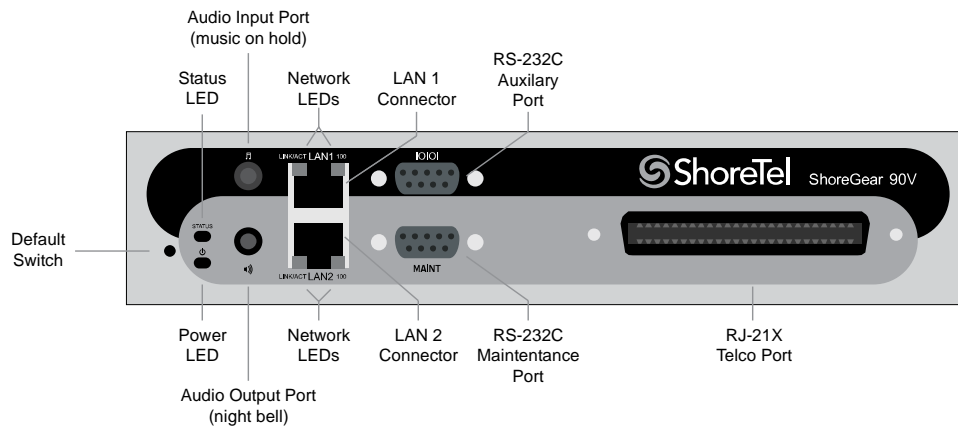


Figure D-12 ShoreGear 90V Front Plate

D.3.1.1 Switch Capacity

- Analog Circuit Resources
 - *Ports 1-8*: Eight Loop Start Trunks
 - *Ports 9-12*: Four Extensions or DID Trunks. A single command configures all ports as either Extensions or DID trunks.
 - *Power Failure Transfer Unit*: Trunk Port 1 to Extension Port 12
- Make Me Conference Resources: 12 ports
 - *Ports 1-12*
- Maximum IP Phone Resources: 90 devices
 - *Analog Port Reallocation*: 60
 - *Built-in Resources*: 30

D.3.1.2 LED Descriptions

Power LED

The ShoreGear 90V has one power LED, which indicates the following:

- On: The switch is operating normally.
- Off: The switch has no power.
- Flashing:
 - *2 flashes*—The switch failed its internal self-test. This indicates a hardware failure. Replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.

- **3 flashes**—Booting via FTP. Flash memory might be corrupted. Go to the **Quick Look** page to ensure that the system is running properly.
- **4 flashes**—The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.
- **5 flashes**—The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreWare server, and set the boot file to `/tsk/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreWare server's IP address, and set option 67 to `/tsk/vxworks`.
- **6 flashes**—Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

The ShoreGear 90V network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:
 - *When flashing*, network activity is detected.
 - *When on (not flashing)*, the switch is connected to an Ethernet network.
 - *When off*, the switch cannot detect an Ethernet network.

This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- **100M:**
 - *When green*, the switch is connected to a 100BaseT network.
 - *When off*, the switch is connected to a 10BaseT network.

Status LED

The ShoreGear 90V has one status LED to provide general information about the ports. The color and blink pattern of the LED indicate the port function:

- **Status LED (Green)**
 - *When on steady*, no ports are handling active calls.
 - *When flashing fast*, at least one port is handling an active call.

- Status LED (Yellow)
 - *When on steady*, no ports are handling active calls and at least one port is out of service.
 - *When flashing slow*, the switch is not connected (or has lost connection) to a ShoreTel server.
 - *When flashing fast*, at least one port is handling an active call and at least one port is out of service.
- Off: No ports are assigned.

D.3.1.3 ShoreGear 90V Connectors

The ShoreGear 90V voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-21X male connector for mass termination of the telephone/trunk ports
 - *Power Failure Transfer Unit*: Trunk Port 1 to Extension Port 12
 - *Backup Operator*: Extension Port 12

ShoreGear 90V RJ-21X Telephone and Trunk Connector

Table D-7 lists the RJ-21X Ring and Tip pin numbers for the ShoreGear 90V

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
1	Trunk	1	Blue/White	26	White/Blue
–		2	Orange/White	27	White/Orange
2	Trunk	3	Green/White	28	White/Green
–		4	Brown/White	29	White/Brown
3	Trunk	5	Slate/White	30	White/Slate
–		6	Blue/Red	31	Red/Blue
4	Trunk	7	Orange/Red	32	Red/Orange
–		8	Green/Red	33	Red/Green
5	Trunk	9	Brown/Red	34	Red/Brown
–		10	Slate/Red	35	Red/Slate
6	Trunk	11	Blue/Black	36	Black/Blue
–		12	Orange/Black	37	Black/Orange
7	Trunk	13	Green/Black	38	Black/Green
–		14	Brown/Black	39	Black/Brown
8	Trunk	15	Slate/Black	40	Black/Slate
–		16	Blue/Yellow	41	Yellow/Blue
9	Extension - DID	17	Orange/Yellow	42	Yellow/Orange
–		18	Green/Yellow	43	Yellow/Green
10	Extension - DID	19	Brown/Yellow	44	Yellow/Brown
–		20	Slate/Yellow	45	Yellow/Slate

Table D-7 ShoreGear 90V RJ-21X Telephone and Trunk Connector Pins

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
11	Extension - DID	21	Blue/Violet	46	Violet/Blue
–		22	Orange/Violet	47	Violet/Orange
12	Extension - DID	23	Green/Violet	48	Violet/Green
–		24	Brown/Violet	49	Violet/Brown
–		25	Slate/Violet	50	Violet/Slate

Table D-7 ShoreGear 90V RJ-21X Telephone and Trunk Connector Pins

D.3.2 ShoreGear 90BRIV Voice Switch

The following sections describe ShoreGear 90BRIV resource capacity, LED behavior, and connectors. Figure D-13 displays the ShoreGear 90BRIV front plate.

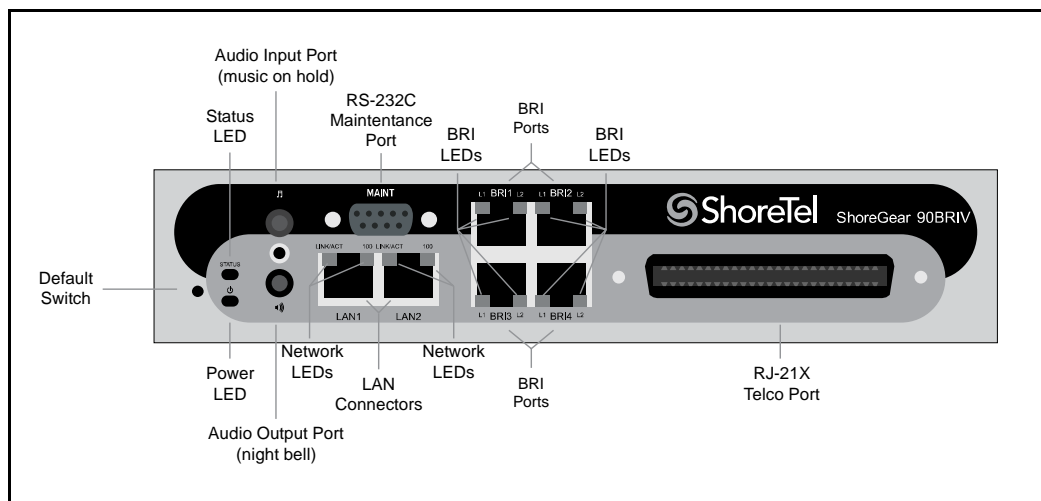


Figure D-13 ShoreGear 90BRIV Front Plate

D.3.2.1 Switch Capacity

- Analog Circuit Resources
 - Ports 9-12: Extensions
- Digital Circuit Resources
 - Four BRI Spans, each comprising two channels: Eight channels maximum
- Make Me Conference Resource: None
- Maximum IP Phone Resources: 90 devices
 - Analog Port Reallocation: 20
 - Digital Channel Reallocation: 40
 - Built-in Resources: 30

D.3.2.2 LED Descriptions

Power LED

The ShoreGear 90BRIV has one power LED, which indicates the following:

- **On:** The switch is operating normally.
- **Off:** The switch has no power.
- **Flashing**
 - **2 flashes**—The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
 - **3 flashes**—Booting via FTP. Flash memory might be corrupted. Go to the **Quick Look** page to ensure that the system is running properly.
 - **4 flashes**—The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.
 - **5 flashes**—The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreWare server, and set the boot file to `/tsk/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreWare server's IP address, and set option 67 to `/tsk/vxworks`.
 - **6 flashes**—Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

The ShoreGear 90BRIV network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:
 - When flashing, network activity is detected.
 - When on (not flashing), the switch is connected to an Ethernet network.
 - When off, the switch cannot detect an Ethernet network.

This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- **100M**
 - *When green*, the switch is connected to a 100BaseT network.
 - *When off*, the switch is connected to a 10BaseT network.

Status LED

The ShoreGear 90BRIV has one status LED to provide general information about the ports. The color and blink pattern of the LED indicate the port function:

- **Status LED (Green)**
 - *When on steady*, no ports are handling active calls.
 - *When flashing fast* (100 msec on/off), at least one port is handling an active call.
- **Status LED (Yellow)**
 - *When on steady*, no ports are handling active calls and at least one port is out of service.
 - *When flashing slow* (1 sec. on/off), the switch is not connected (or has lost connection) to a ShoreTel server.
 - *When flashing fast* (100 msec on/off), at least one port is handling an active call and at least one port is out of service.
- **Off**: No ports are assigned.

BRI LED

Each BRI connector has two LEDs to indicate port activity. The color and blink pattern of the LED indicate the port function:

- LED 1: Off, LED 2 Off – Port not configured in Director
- LED 1: Yellow, LED 2 Off – Port inactive or not connected
- LED 1: Off, LED 2 Off – Layer 1 active. Layer 2 not established
- LED 1: Off, LED 2 Green – Layer 1 active. Layer 2 active.
- LED 1: Off, LED 2 Green flashing – Call in progress (Layer 1, Layer 2, and Layer 3 active).

D.3.2.3 ShoreGear 90BRIV Connectors

The ShoreGear 90BRIV voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-21X male connector for mass termination of the telephone/trunk ports
- 4 RJ-45 T1 telco port

ShoreGear 90BRIV RJ-21X Telephone and Trunk Connector

Table D-8 lists the RJ-21X Ring and Tip pin numbers for the SG 90BRIV

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
–		1	Blue/White	26	White/Blue
–		2	Orange/White	27	White/Orange
–		3	Green/White	28	White/Green
–		4	Brown/White	29	White/Brown
–		5	Slate/White	30	White/Slate
–		6	Blue/Red	31	Red/Blue

Table D-8 ShoreGear 90BRIV RJ-21X Telephone and Trunk Connector Pins

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
–		7	Orange/Red	32	Red/Orange
–		8	Green/Red	33	Red/Green
–		9	Brown/Red	34	Red/Brown
–		10	Slate/Red	35	Red/Slate
–		11	Blue/Black	36	Black/Blue
–		12	Orange/Black	37	Black/Orange
–		13	Green/Black	38	Black/Green
–		14	Brown/Black	39	Black/Brown
–		15	Slate/Black	40	Black/Slate
–		16	Blue/Yellow	41	Yellow/Blue
9	Extension	17	Orange/Yellow	42	Yellow/Orange
–		18	Green/Yellow	43	Yellow/Green
10	Extension	19	Brown/Yellow	44	Yellow/Brown
–		20	Slate/Yellow	45	Yellow/Slate
11	Extension	21	Blue/Violet	46	Violet/Blue
–		22	Orange/Violet	47	Violet/Orange
12	Extension	23	Green/Violet	48	Violet/Green
–		24	Brown/Violet	49	Violet/Brown
–		25	Slate/Violet	50	Violet/Slate

Table D-8 ShoreGear 90BRIV RJ-21X Telephone and Trunk Connector Pins

D.3.3 ShoreGear 50V Voice Switch

The following sections describe ShoreGear 50V resource capacity, LED behavior, and connectors. The ShoreGear 50V is not supported in installations outside the U.S. and Canada. Figure D-14 displays the ShoreGear 50V front plate.

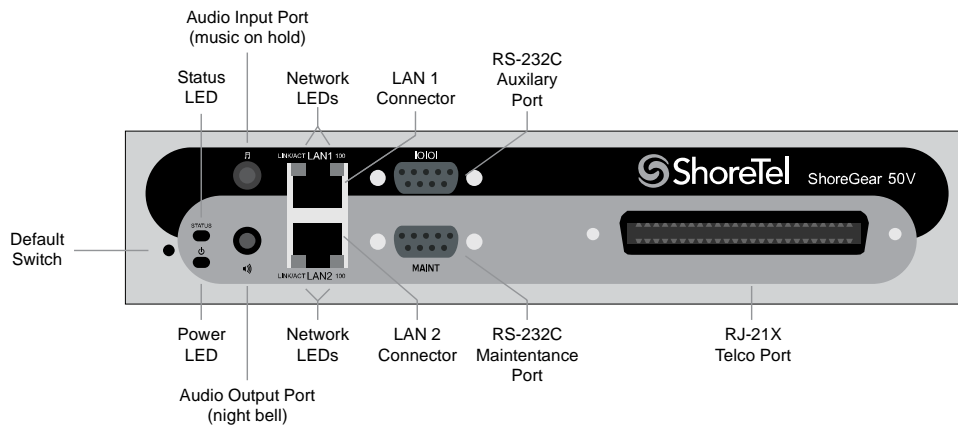


Figure D-14 ShoreGear 50V Front Plate

D.3.3.1 Switch Capacity

- Analog Circuit Resources
 - *Ports 1-4*: Four Loop Start Trunks
 - *Ports 11-12*: Two Extensions or DID Trunks. A single command configures all ports as either Extensions or DID trunks.
 - *Power Failure Transfer Unit*: Trunk Port 1 to Extension Port 12
- Make Me Conference Resources: six ports
 - *Ports 1-4, 11-12*
- Maximum IP Phone Resources: 50 devices
 - *Analog Port Reallocation*: 30
 - *Built-in Resources*: 20

D.3.3.2 LED Descriptions

Power LED

The ShoreGear 50V has one power LED, which indicates the following:

- On: The switch is operating normally.
- Off: The switch has no power.
- Flashing
 - *2 flashes*—The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
 - *3 flashes*—Bootting via FTP. Flash memory might be corrupted. Go to the **Quick Look** page to ensure that the system is running properly.
 - *4 flashes*—The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.
 - *5 flashes*—The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreWare server, and set the boot file to `/tsk/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreWare server's IP address, and set option 67 to `/tsk/vxworks`.
 - *6 flashes*—Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

The ShoreGear 50V network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:
 - *When flashing*, network activity is detected.
 - *When on (not flashing)*, the switch is connected to an Ethernet network.
 - *When off*, the switch cannot detect an Ethernet network.

This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- **100M**
 - *When green*, the switch is connected to a 100BaseT network.
 - *When off*, the switch is connected to a 10BaseT network.

Status LED

The ShoreGear 50V has one status LED to provide general information about the ports. The color and blink pattern of the LED indicate the port function:

- **Status LED (Green)**
 - *When on steady*, no ports are handling active calls.
 - *When flashing fast*, at least one port is handling an active call.
- **Status LED (Yellow)**
 - *When on steady*, no ports are handling active calls and at least one port is out of service.
 - *When flashing slow*, the switch is not connected (or has lost connection) to a ShoreTel server.
 - *When flashing fast*, at least one port is handling an active call and at least one port is out of service.
- **Off:** No ports are assigned.

D.3.3.3 ShoreGear 50V Connectors

The ShoreGear 50V voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-21X male connector for mass termination of the telephone/trunk ports
 - *Power Failure Transfer Unit:* Trunk Port 1 to Extension Port 12
 - *Backup Operator:* Extension Port 12

ShoreGear 50V RJ-21X Telephone and Trunk Connector

Table D-9 lists the RJ-21X Ring and Tip pin numbers for the SG 50V

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
1	Trunk	1	Blue/White	26	White/Blue
–		2	Orange/White	27	White/Orange
2	Trunk	3	Green/White	28	White/Green
–		4	Brown/White	29	White/Brown
3	Trunk	5	Slate/White	30	White/Slate
–		6	Blue/Red	31	Red/Blue
4	Trunk	7	Orange/Red	32	Red/Orange
–		8	Green/Red	33	Red/Green
–		9	Brown/Red	34	Red/Brown
–		10	Slate/Red	35	Red/Slate
–		11	Blue/Black	36	Black/Blue
–		12	Orange/Black	37	Black/Orange
–		13	Green/Black	38	Black/Green
–		14	Brown/Black	39	Black/Brown
–		15	Slate/Black	40	Black/Slate
–		16	Blue/Yellow	41	Yellow/Blue
9	Extension - DID	17	Orange/Yellow	42	Yellow/Orange
–		18	Green/Yellow	43	Yellow/Green
10	Extension - DID	19	Brown/Yellow	44	Yellow/Brown
–		20	Slate/Yellow	45	Yellow/Slate
11	Extension - DID	21	Blue/Violet	46	Violet/Blue
–		22	Orange/Violet	47	Violet/Orange
12	Extension - DID	23	Green/Violet	48	Violet/Green
–		24	Brown/Violet	49	Violet/Brown
–		25	Slate/Violet	50	Violet/Slate

Table D-9 ShoreGear 50V RJ-21X Telephone and Trunk Connector Pins

D.4 Specification – SG 1U Full Width Switches

D.4.1 ShoreGear 120 Voice Switch

The following sections describe ShoreGear 120 resource capacity, LED behavior, and connectors. The ShoreGear 120 is not supported in installations outside the U.S. and Canada. Figure D-15 displays the ShoreGear 120 front plate.

The ShoreGear 120 is often referred to as the ShoreGear 120/24 (SG 120/24).

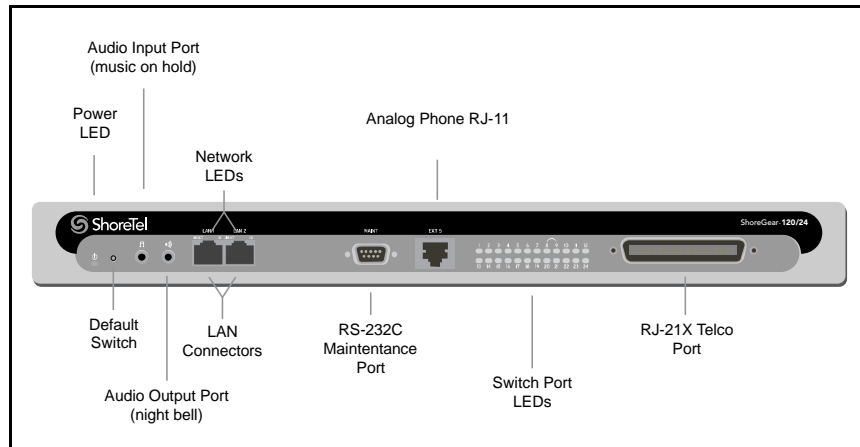


Figure D-15 ShoreGear 120 Front Plate

D.4.1.1 Switch Capacity

- Analog Circuit Resources
 - *Ports 1-8*: Eight Loop Start Trunks, DID Trunks, or Extensions
 - *Ports 9-24*: Sixteen Extensions.
 - *Power Failure Transfer Unit*: Trunk Port 8 to Extension Port 9
- Make Me Conference Resource: 24 Ports
 - *Ports 1-24*
- Maximum IP Phone Resources: 120 devices
 - *Analog Port Reallocation*: 120

D.4.1.2 LED Descriptions

Power LED

The ShoreGear 120 has one power LED, which indicates the following:

- On: The switch is operating normally.
- Off: The switch has no power.
- Flashing
 - *2 flashes*—The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.

- **3 flashes**—Booting via FTP. Flash memory might be corrupted. Go to the **Quick Look** page to ensure that the system is running properly.
- **4 flashes**—The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.
- **5 flashes**—The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreWare server, and set the boot file to `/tsa/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreWare server's IP address, and set option 67 to `/tsa/vxworks`.
- **6 flashes**—Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Switch Port LEDs

The ShoreGear 120 has 24 telephone/trunk port LEDs. The color of the LED indicates the port function:

- **Green** when the port is a telephone port.
- **Yellow** when the port is a trunk port.
- **Off** indicates the port is reserved for IP phones, for conferencing, or is unconfigured.

The following describes the switch port LED behavior and meaning:

- **Telephone Port LEDs (Green)**
 - *When on steady*, the port is configured as a telephone port and the telephone is idle.
 - *When flashing with ring cadence*, the telephone is ringing.
 - *When flashing slowly*, the telephone is off hook.
 - *When flashing fast*, the port is in use (call in progress).
- **Trunk Port LED (Yellow):**
 - *When on steady*, the port is configured as a trunk port and the trunk is idle.
 - *When flashing slowly*, the trunk is off hook.
 - *When flashing fast*, the trunk is in use (call in progress).
- **Port LED Alternating Green/Yellow:** The port is out of service. The LED periodically alternates green/yellow or yellow/green. The color of the LED between alternating colors indicates the port type: green for phone and yellow for trunk.
- **Off (IP phone):** When the LED is off, the port is reserved for IP phones, for conferencing, or is unconfigured.

Network LEDs

The network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:
 - *When flashing*, network activity is detected.
 - *When on* (not flashing), heavy network activity is detected.
 - *When off*, the switch has no power.

This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- **100M**
 - *When green*, the switch is connected to a 100BaseT network.
 - *When off*, the switch is connected to a 10BaseT network.

D.4.1.3 ShoreGear 120 Connectors

The ShoreGear 120 voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-11 connector for connecting an analog phone (extension 9)
- 1 RJ-21X male connector for mass termination of the telephone/trunk ports
 - *Power Failure Transfer Unit:* Trunk Port 8 to Extension Port 9
 - *Backup Operator:* Extension Port 9

ShoreGear 120 RJ-21X Telephone and Trunk Connector

Table D-10 lists the RJ-21X Ring and Tip pin numbers for the SG 120I

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
1	Trunk, DID, Extension	1	Blue/White	26	White/Blue
2	Trunk, DID, Extension	2	Orange/White	27	White/Orange
3	Trunk, DID, Extension	3	Green/White	28	White/Green
4	Trunk, DID, Extension	4	Brown/White	29	White/Brown
5	Trunk, DID, Extension	5	Slate/White	30	White/Slate
6	Trunk, DID, Extension	6	Blue/Red	31	Red/Blue
7	Trunk, DID, Extension	7	Orange/Red	32	Red/Orange
8	Trunk, DID, Extension	8	Green/Red	33	Red/Green
9	Extension	9	Brown/Red	34	Red/Brown

Table D-10 ShoreGear 120 RJ-21X Telephone and Trunk Connector Pins

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
10	Extension	10	Slate/Red	35	Red/Slate
11	Extension	11	Blue/Black	36	Black/Blue
12	Extension	12	Orange/Black	37	Black/Orange
13	Extension	13	Green/Black	38	Black/Green
14	Extension	14	Brown/Black	39	Black/Brown
15	Extension	15	Slate/Black	40	Black/Slate
16	Extension	16	Blue/Yellow	41	Yellow/Blue
17	Extension	17	Orange/Yellow	42	Yellow/Orange
18	Extension	18	Green/Yellow	43	Yellow/Green
19	Extension	19	Brown/Yellow	44	Yellow/Brown
20	Extension	20	Slate/Yellow	45	Yellow/Slate
21	Extension	21	Blue/Violet	46	Violet/Blue
22	Extension	22	Orange/Violet	47	Violet/Orange
23	Extension	23	Green/Violet	48	Violet/Green
24	Extension	24	Brown/Violet	49	Violet/Brown
–		25	Slate/Violet	50	Violet/Slate

Table D-10 ShoreGear 120 RJ-21X Telephone and Trunk Connector Pins

D.4.2 ShoreGear 24A Voice Switch

The following sections describe ShoreGear 24A resource capacity, LED behavior, and connectors. The ShoreGear 24A is not supported in installations outside the U.S. and Canada. Figure D-16 displays the ShoreGear 24A front plate.

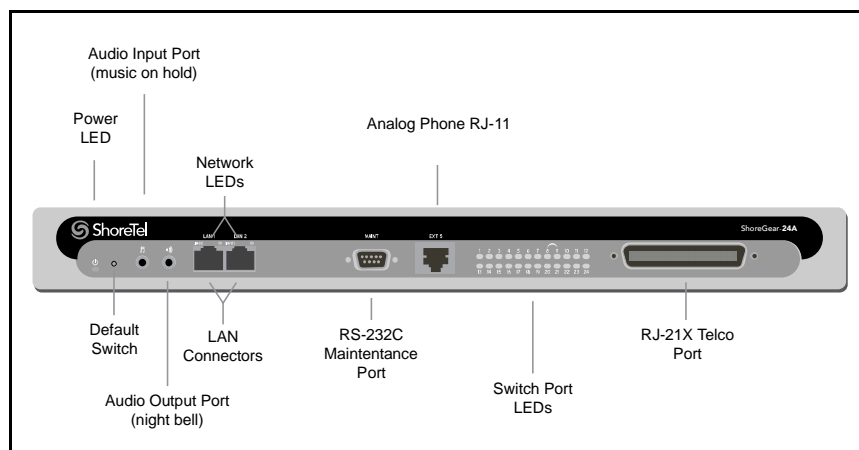


Figure D-16 ShoreGear 24A Front Plate

D.4.2.1 Switch Capacity

- Analog Circuit Resources
 - Ports 1-24: Twenty four extensions
- Make Me Conference Resource: 24 Ports

- Ports 1-24
- Maximum IP Phone Resources: None

D.4.2.2 LED Descriptions

Power LED

The ShoreGear 24A has one power LED, which indicates the following:

- **On:** The switch is operating normally.
- **Off:** The switch has no power.
- **Flashing**
 - **2 flashes**—The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
 - **3 flashes**—Booting via FTP. Flash memory might be corrupted. Go to the **Quick Look** page to ensure that the system is running properly.
 - **4 flashes**—The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.
 - **5 flashes**—The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreWare server, and set the boot file to `/tsa/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreWare server's IP address, and set option 67 to `/tsa/vxworks`.
 - **6 flashes**—Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Switch Port LEDs

The ShoreGear 24A has 24 telephone/trunk port LEDs. The color of the LED indicates the port function:

- **Green** when the port is a telephone port.
- **Yellow** when the port is a trunk port.
- **Off** indicates the port is reserved for IP phones, for conferencing, or is unconfigured.

The following describes the switch port LED behavior and meaning:

- **Telephone Port LEDs (Green)**
 - *When on steady*, the port is configured as a telephone port and the telephone is idle.

- *When flashing with ring cadence*, the telephone is ringing.
- *When flashing slowly*, the telephone is off hook.
- *When flashing fast*, the port is in use (call in progress).
- **Trunk Port LED (Yellow):**
 - *When on steady*, the port is configured as a trunk port and the trunk is idle.
 - *When flashing slowly*, the trunk is off hook.
 - *When flashing fast*, the trunk is in use (call in progress).
- **Port LED Alternating Green/Yellow:** The port is out of service. The LED periodically alternates green/yellow or yellow/green. The color of the LED between alternating colors indicates the port type: green for phone and yellow for trunk.
- **Off (IP phone):** When the LED is off, the port is reserved for IP phones, for conferencing, or is unconfigured.

Network LEDs

The network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:
 - *When flashing*, network activity is detected.
 - *When on* (not flashing), heavy network activity is detected.
 - *When off*, the switch has no power.

This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- **100M**
 - *When green*, the switch is connected to a 100BaseT network.
 - *When off*, the switch is connected to a 10BaseT network.

D.4.2.3 ShoreGear 24A Connectors

The ShoreGear 24A voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-11 connector for connecting an analog phone (extension 9)
- 1 RJ-21X male connector for mass termination of the telephone/trunk ports

ShoreGear 24A RJ-21X Telephone and Trunk Connector

Table D-11 lists the RJ-21X Ring and Tip pin numbers for the SG 24AI

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
1	Extension	1	Blue/White	26	White/Blue
2	Extension	2	Orange/White	27	White/Orange
3	Extension	3	Green/White	28	White/Green
4	Extension	4	Brown/White	29	White/Brown
5	Extension	5	Slate/White	30	White/Slate
6	Extension	6	Blue/Red	31	Red/Blue
7	Extension	7	Orange/Red	32	Red/Orange
8	Extension	8	Green/Red	33	Red/Green
9	Extension	9	Brown/Red	34	Red/Brown
10	Extension	10	Slate/Red	35	Red/Slate
11	Extension	11	Blue/Black	36	Black/Blue
12	Extension	12	Orange/Black	37	Black/Orange
13	Extension	13	Green/Black	38	Black/Green
14	Extension	14	Brown/Black	39	Black/Brown
15	Extension	15	Slate/Black	40	Black/Slate
16	Extension	16	Blue/Yellow	41	Yellow/Blue
17	Extension	17	Orange/Yellow	42	Yellow/Orange
18	Extension	18	Green/Yellow	43	Yellow/Green
19	Extension	19	Brown/Yellow	44	Yellow/Brown
20	Extension	20	Slate/Yellow	45	Yellow/Slate
21	Extension	21	Blue/Violet	46	Violet/Blue
22	Extension	22	Orange/Violet	47	Violet/Orange
23	Extension	23	Green/Violet	48	Violet/Green
24	Extension	24	Brown/Violet	49	Violet/Brown
–		25	Slate/Violet	50	Violet/Slate

Table D-11 ShoreGear 24A RJ-21X Telephone and Trunk Connector Pins

D.4.3 ShoreGear 60 Voice Switch

The following sections describe ShoreGear 60 resource capacity, LED behavior, and connectors. The ShoreGear 60 is not supported in installations outside the U.S. and Canada. Figure D-17 displays the ShoreGear 60 front plate.

The ShoreGear 60 is often referred to as the ShoreGear 60/12 (SG 60/12).

D.4.3.1 Switch Capacity

- Analog Circuit Resources
 - Ports 1-8: Eight Loop Start Trunks, DID Trunks, or Extensions
 - Ports 9-12: Four Extensions.
 - Backup Operator: Extension Port 9
 - Power Failure Transfer Unit: Trunk Port 8 to Extension Port 9
- Make Me Conference Resource: 12 ports

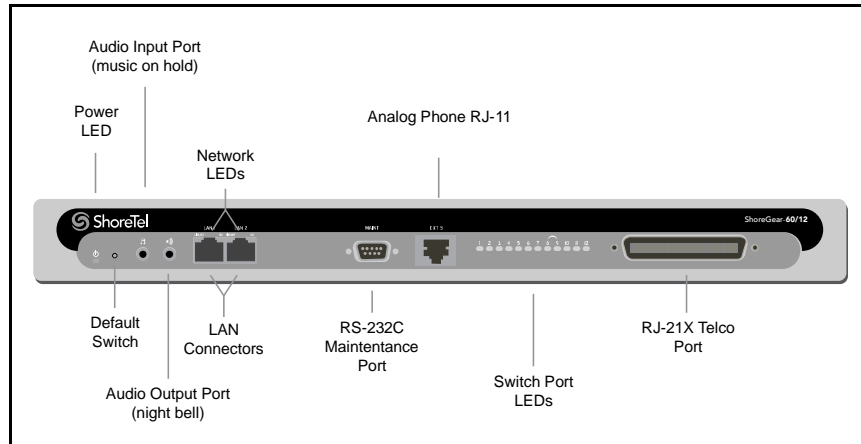


Figure D-17 ShoreGear 60 Front Plate

— *Ports 1-12*

- **Maximum IP Phone Resources:** 60 devices
- *Analog Port Reallocation:* 60

D.4.3.2 LED Descriptions

Power LED

The ShoreGear 60 has one power LED, which indicates the following:

- **On:** The switch is operating normally.
- **Off:** The switch has no power.
- **Flashing**
 - **2 flashes**—The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
 - **3 flashes**—Booting via FTP. Flash memory might be corrupted. Go to the **Quick Look** page to ensure that the system is running properly.
 - **4 flashes**—The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.
 - **5 flashes**—The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreWare server, and set the boot file to `/tsa/vxworks`. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreWare server's IP address, and set option 67 to `/tsa/vxworks`.
 - **6 flashes**—Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using

the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Switch Port LEDs

The ShoreGear 60 has 12 telephone/trunk port LEDs. The color of the LED indicates the port function:

- **Green** when the port is a telephone port.
- **Yellow** when the port is a trunk port.
- **Off** indicates the port is reserved for IP phones, for conferencing, or is unconfigured.

The following describes the switch port LED behavior and meaning:

- **Telephone Port LEDs (Green)**
 - *When on steady*, the port is configured as a telephone port and the telephone is idle.
 - *When flashing with ring cadence*, the telephone is ringing.
 - *When flashing slowly*, the telephone is off hook.
 - *When flashing fast*, the port is in use (call in progress).
- **Trunk Port LED (Yellow)**
 - *When on steady*, the port is configured as a trunk port and the trunk is idle.
 - *When flashing slowly*, the trunk is off hook.
 - *When flashing fast*, the trunk is in use (call in progress).
- **Port LED Alternating Green/Yellow:** The port is out of service. The LED periodically alternates green/yellow or yellow/green. The LED color between alternating colors indicates the port type: green for phone and yellow for trunk.
- **Off (IP phone):** When the LED is off, the port is reserved for IP phones, for conferencing, or is unconfigured.

Network LEDs

The network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:
 - *When flashing*, network activity is detected.
 - *When on* (not flashing), heavy network activity is detected.
 - *When off*, the switch has no power.

This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- **100M**
 - *When green*, the switch is connected to a 100BaseT network.
 - *When off*, the switch is connected to a 10BaseT network.

D.4.3.3 ShoreGear 60 Connectors

ShoreGear 60 voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-11 connector for connecting an analog phone (extension 9)
- 1 RJ-21X male connector for mass termination of the telephone/trunk ports
 - *Power Failure Transfer Unit*: Trunk Port 8 to Extension Port 9
 - *Backup Operator*: Extension Port 9

ShoreGear 60 RJ-21X Telephone and Trunk Connector

Table D-12 lists the RJ-21X Ring and Tip pin numbers for the SG 60.

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
1	Trunk, DID, Extension	1	Blue/White	26	White/Blue
2	Trunk, DID, Extension	2	Orange/White	27	White/Orange
3	Trunk, DID, Extension	3	Green/White	28	White/Green
4	Trunk, DID, Extension	4	Brown/White	29	White/Brown
5	Trunk, DID, Extension	5	Slate/White	30	White/Slate
6	Trunk, DID, Extension	6	Blue/Red	31	Red/Blue
7	Trunk, DID, Extension	7	Orange/Red	32	Red/Orange
8	Trunk, DID, Extension	8	Green/Red	33	Red/Green
9	Extension	9	Brown/Red	34	Red/Brown
10	Extension	10	Slate/Red	35	Red/Slate
11	Extension	11	Blue/Black	36	Black/Blue
12	Extension	12	Orange/Black	37	Black/Orange
–		13	Green/Black	38	Black/Green
–		14	Brown/Black	39	Black/Brown
–		15	Slate/Black	40	Black/Slate
–		16	Blue/Yellow	41	Yellow/Blue
–		17	Orange/Yellow	42	Yellow/Orange
–		18	Green/Yellow	43	Yellow/Green
–		19	Brown/Yellow	44	Yellow/Brown
–		20	Slate/Yellow	45	Yellow/Slate
–		21	Blue/Violet	46	Violet/Blue
–		22	Orange/Violet	47	Violet/Orange
–		23	Green/Violet	48	Violet/Green
–		24	Brown/Violet	49	Violet/Brown
–		25	Slate/Violet	50	Violet/Slate

Table D-12 ShoreGear 60 RJ-21X Telephone and Trunk Connector Pins

D.4.3.4 ShoreGear 40 Voice Switch

The following sections describe ShoreGear 40 resource capacity, LED behavior, and connectors. The ShoreGear 40 is not supported in installations outside the U.S. and Canada. Figure D-18 displays the ShoreGear 40 front plate.

The ShoreGear 40 is often referred to as the ShoreGear 40/8 (SG 40/8)

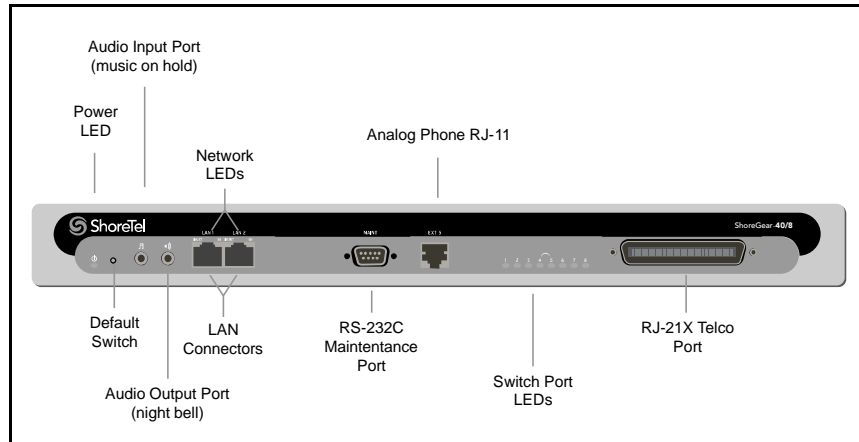


Figure D-18 ShoreGear 40 Front Plate

D.4.3.5 Switch Capacity

- Analog Circuit Resources
 - *Ports 1-2*: Two Loop Start Trunks, DID Trunks, or Extensions
 - *Ports 3-4*: Two Loop Start Trunks.
 - *Ports 5-8*: Four Extensions
 - *Power Failure Transfer Unit*: Trunk Port 4 to Extension Port 5
- Make Me Conference Resource: eight ports
 - *Ports 1-8*
- Maximum IP Phone Resources: 40 devices
 - *Analog Port Reallocation*: 40

D.4.3.6 LED Descriptions

Power LED

The ShoreGear 40 has one power LED, which indicates the following:

- On: The switch is operating normally.
- Off: The switch has no power.
- Flashing
 - *2 flashes*—The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
 - *3 flashes*—Booting via FTP. Flash memory might be corrupted. Go to the **Quick Look** page to ensure that the system is running properly.

- **4 flashes**—The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.
- **5 flashes**—The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreWare server, and set the boot file to /tsa/vxworks. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreWare server's IP address, and set option 67 to /tsa/vxworks.
- **6 flashes**—Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Switch Port LEDs

The ShoreGear 40 has 8 telephone/trunk port LEDs. The color of the LED indicates the port function:

- **Green** when the port is a telephone port.
- **Yellow** when the port is a trunk port.
- **Off** indicates the port is reserved for IP phones, for conferencing, or is unconfigured.

The following describes the switch port LED behavior and meaning.

- **Telephone Port LEDs (Green)**
 - *When on steady*, the port is configured as a telephone port and the telephone is idle.
 - *When flashing with ring cadence*, the telephone is ringing.
 - *When flashing slowly*, the telephone is off hook.
 - *When flashing fast*, the port is in use (call in progress).
- **Trunk Port LED (Yellow)**
 - *When on steady*, the port is configured as a trunk port and the trunk is idle.
 - *When flashing slowly*, the trunk is off hook.
 - *When flashing fast*, the trunk is in use (call in progress).
- **Port LED Alternating Green/Yellow:** The port is out of service. The LED periodically alternates green/yellow or yellow/green. The color of the LED between alternating colors indicates the port type: green for phone and yellow for trunk.
- **Off (IP phone):** When the LED is off, the port is reserved for IP phones, for conferencing, or is unconfigured.

Network LEDs

The network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity.

When both LAN connectors are connected into a redundant network configuration, one network port is active while the other is in standby mode. If one LAN connection fails, the switch activates the other port.

The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:
 - *When flashing*, network activity is detected.
 - *When on* (not flashing), heavy network activity is detected.
 - *When off*, network activity is not detected.

This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- **100M**
 - *When green*, the switch is connected to a 100BaseT network.
 - *When off*, the switch is connected to a 10BaseT network.

D.4.3.7 ShoreGear 40 Connectors

The ShoreGear 40 voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-11 connector for connecting an analog phone (extension 9)
- 1 RJ-21X male connector for mass termination of the telephone/trunk ports
 - *Power Failure Transfer Unit:* Trunk Port 4 to Extension Port 5
 - *Backup Operator:* Extension Port 5

ShoreGear 40 RJ-21X Telephone and Trunk Connector

Table D-13 lists the RJ-21X Ring and Tip pin numbers for the SG 40I

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
1	Trunk, DID, Extension	1	Blue/White	26	White/Blue
2	Trunk, DID, Extension	2	Orange/White	27	White/Orange
3	Trunk	3	Green/White	28	White/Green
4	Trunk	4	Brown/White	29	White/Brown
5	Extension	5	Slate/White	30	White/Slate
6	Extension	6	Blue/Red	31	Red/Blue
7	Extension	7	Orange/Red	32	Red/Orange
8	Extension	8	Green/Red	33	Red/Green
–		9	Brown/Red	34	Red/Brown
–		10	Slate/Red	35	Red/Slate
–		11	Blue/Black	36	Black/Blue
–		12	Orange/Black	37	Black/Orange

Table D-13 ShoreGear 40 RJ-21X Telephone and Trunk Connector Pins

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
–		13	Green/Black	38	Black/Green
–		14	Brown/Black	39	Black/Brown
–		15	Slate/Black	40	Black/Slate
–		16	Blue/Yellow	41	Yellow/Blue
–		17	Orange/Yellow	42	Yellow/Orange
–		18	Green/Yellow	43	Yellow/Green
–		19	Brown/Yellow	44	Yellow/Brown
–		20	Slate/Yellow	45	Yellow/Slate
–		21	Blue/Violet	46	Violet/Blue
–		22	Orange/Violet	47	Violet/Orange
–		23	Green/Violet	48	Violet/Green
–		24	Brown/Violet	49	Violet/Brown
–		25	Slate/Violet	50	Violet/Slate

Table D-13 ShoreGear 40 RJ-21X Telephone and Trunk Connector Pins

D.4.4 ShoreGear-T1 and ShoreGear E1 Voice Switches

The following sections describe ShoreGear-T1 and ShoreGear E1 resource capacity, LED behavior, and connectors. The ShoreGear-T1 is not supported in installations outside the U.S. and Canada. Figure D-19 displays the ShoreGear-T1 front plate. The ShoreGear E1 front plate is identical to the ShoreGear-T1 except for the E1 labeling.

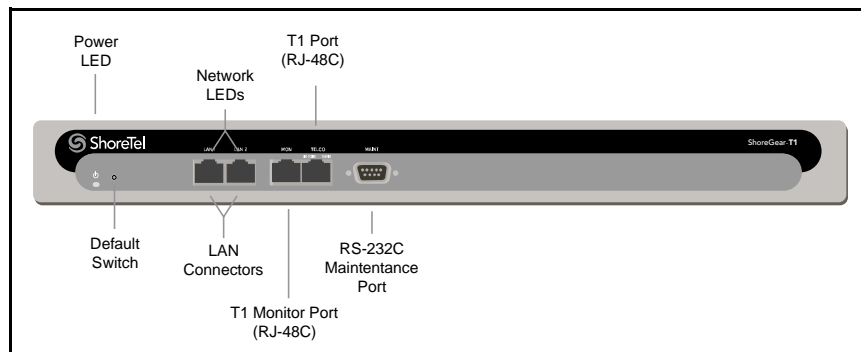


Figure D-19 ShoreGear-T1 Front Plate

The ShoreGear-T1 provides higher-density trunking to the central office using CAS or PRI signaling. The ShoreGear-T1 can also be used as a gateway to legacy PBX systems.

The ShoreGear E1 provides higher-density trunking to the central office using PRI signaling. The ShoreGear E1 can also be used as a gateway to legacy PBX systems.

D.4.4.1 Switch Capacity

- Digital Circuit Resources
 - SG T1 – One T1 circuit, 24 channels per circuit: 24 channels maximum.
 - SG E1 – One T1 circuit, 30 channels per circuit: 30 channels maximum.

- Make Me Conference Resources: None
- Maximum IP Phone Resources: None

D.4.4.2 LED Descriptions

Power LED

The ShoreGear-T1 and ShoreGear E1 voice switches have one power LED, which indicates the following:

- **On:** The switch is operating normally.
- **Off:** The switch has no power.
- **Flashing**
 - **2 flashes**—The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
 - **3 flashes**—Booting via FTP. Flash memory may be corrupted. Go to the **Quick Look** page to make sure that the system is running properly.
 - **4 flashes**—The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.
 - **5 flashes**—The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds.
 - **6 flashes**—Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

The network LEDs (LAN1 and LAN2) indicate the speed at which the switch is communicating with the network and whether there is network activity. The network LED descriptions are as follows:

- **Link/Activity:** When lit, this LED indicates that the switch is connected to an Ethernet network. This LED indicates network activity, as follows:
 - **When flashing**, network activity is detected.
 - **When on** (not flashing), heavy network activity is detected.
 - **When off**, network activity is not detected.

This LED is not directly related to any switch's individual network activity. For example, if three switches are connected to the same hub and one switch's Traffic LED shows activity, the other switches will indicate the same activity.

- **100M**
 - **When green**, the switch is connected to a 100BaseT network.

- *When off*, the switch is connected to a 10BaseT network.

Monitor and Telco LEDs

The Monitor and Telco LEDs indicate line coding, network framing, and loopback status. These LEDs are color coded—green, yellow, and red. The Monitor and Telco LED descriptions follow.

Telco and Monitor LED alarms and errors are logged as switch events in ShoreWare Director's event log.

- **Line Coding:** This LED indicates line coding status, as follows:
 - *When green*, the line coding signal is good.
 - *When yellow*, bipolar violations (BPV) are being received at one second intervals.
 - *When red*, a loss of signal (LOS) has occurred.
 - *When flashing red*, loopback is active (local or CO).
 - *When off*, the switch has no power.
- **Framing:** This LED indicates network framing status, as follows:
 - *When green*, the T1/E1 signal is *in frame*; the signal is synchronized.
 - *When yellow*, a *yellow alarm* has been received from the Central Office.
 - *When flashing yellow*, the frame-bit error rate has exceeded its limits.
 - *When flashing slow yellow*, a small number of frame-bit errors (>1 per million) have occurred; this state will take up to 10 minutes to clear.
 - *When flashing fast yellow*, a series of frame-bit errors (>1 per 1000) have occurred.
 - *When red*, the T1/E1 signal is out-of-frame (OOF). The received signal cannot be framed to the Extended Superframe (ESF) or D4 format.
 - *When flashing red*, loopback is active (local or CO).
 - *When off*, the switch has no power.

ShoreGear-T1 and ShoreGear E1 Connectors

The ShoreGear-T1 and ShoreGear E1 voice switches contain the following components:

- 1 DB-9 female connector for maintenance
- 2 RJ-45 connectors for the LAN interface
- 1 RJ-45 T1 telco port
- 1 RJ-45 T1 monitor port for connecting test equipment

D.5 Specification – IPBX Switches

D.5.1 IPBX-24 Voice Switch

The IPBX-24 (Figure D-20) supports 24 analog ports in a combination of no more than 8 standard trunk ports and 16 analog telephone ports, up to 24. The IPBX-24 interfaces to standard analog trunks using loop start or wink start signaling, as well as to standard analog telephones, including CLASS feature phones with Caller ID Name and Number, and Message Waiting.

The IPBX-24 can be configured to support up to 120 ShoreTel IP phones (five IP phones per IP phone port). The IPBX-24 ports are configured for IP through the *Edit ShoreGear 24* page.

If you are installing IP phones in your system, you must assign sufficient IPBX-24 ports to IP phone support before attempting to install and configure IP phones.

The IPBX-24 does not support analog trunks outside the U.S. and Canada. The IPBX-24 does not support Caller ID, Message Waiting, Fax, and modems outside the U.S. and Canada.

The IPBX-24 LEDs and connectors are defined below.

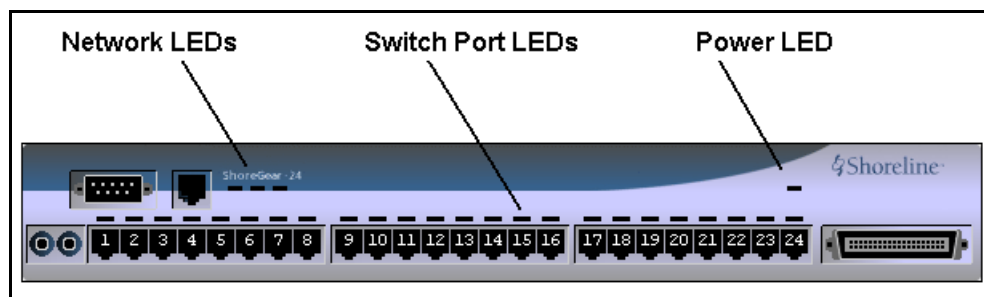


Figure D-20 IPBX-24 Front Plate

D.5.1.1 LED Descriptions

Power LED

The IPBX-24 has one green power LED, which indicates the following:

- **On:** The switch is operating normally.
- **Off:** The switch has no power.
- **Flashing**
 - *2 flashes*—The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
 - *3 flashes*—Booting via FTP. Flash memory might be corrupted. Go to the **Quick Look** page to ensure that the system is running properly.
 - *4 flashes*—The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.

- *5 flashes*—The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds. You can use BOOTP or DHCP to tell the switch where the files are. If you are using BOOTP, set the BOOTP server to the IP address of the ShoreWare server, and set the boot file to /ts24/vxworks. If you are using a DHCP server that supports options 66 and 67, set option 66 to the ShoreWare server's IP address, and set option 67 to /ts24/vxworks.
- *6 flashes*—Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Switch Port LEDs

The IPBX-24 has 24 telephone/trunk port LEDs. The color of the LED indicates the port function:

- **Green** when the port is a telephone port.
- **Yellow** when the port is a trunk port.
- **Off** indicates the port is reserved for IP phones, for conferencing, or is unconfigured.

The following describes the switch port LED behavior and meaning:

- **Telephone Port LEDs (Green)**
 - *When on steady*, the port is configured as a telephone port and the telephone is idle.
 - *When flashing with ring cadence*, the telephone is ringing.
 - *When flashing slowly*, the telephone is off hook.
 - *When flashing fast*, the port is in use (call in progress).
- **Trunk Port LED (Yellow)**
 - *When on steady*, the port is configured as a trunk port and the trunk is idle.
 - *When flashing slowly*, the trunk is off hook.
 - *When flashing fast*, the trunk is in use (call in progress).
- **Port LED (Red):** When on, the port is out of service.
- **IP Phone (Off):** When the LED is off, the port is reserved for IP phones, for conferencing, or is unconfigured.

Network LEDs

The network LEDs indicate the speed at which the switch is communicating with the network and whether there is network activity.

The network LED descriptions are as follows:

- **Link:** When lit, the switch is connected to an Ethernet network.
- **100M**
 - *When lit*, the switch is connected to a 100BaseT network.
 - *When off*, the switch is connected to a 10BaseT network.
- **Traffic:** This LED indicates network activity, as follows:

- *When flashing*, network activity is detected.
- *When on (not flashing)*, heavy network activity is detected.
- *When off*, network activity is not detected.

This LED is not directly related to any particular switch's individual network activity. For example, if you have three switches that are connected to the same hub, and one switch's Traffic LED shows activity, the other switches will indicate the same.

D.5.1.2 IPBX-24 Connectors

The IPBX-24 voice switch contains the following components:

- 1 3.5 mm mono connector for audio input (music on hold)
- 1 3.5 mm mono connector for audio output (overhead paging and night bell)
- 1 DB-9 female connector for maintenance
- 1 RJ-45 connector for the LAN interface
- 24 RJ-11 connectors for the trunk and telephone ports:
 - 8 universal telephone/trunk ports (Ports 1 through 8)
 - 16 telephone ports (Ports 9 through 24)
 - Power fail transfer (between Ports 8 and 9)
- 1 RJ-21X male connector for mass termination of the telephone/trunk ports

IPBX-24 RJ-21X Telephone and Trunk Connector

Table D-14 lists the RJ-21X Ring and Tip pin numbers for the IPBX-24I

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
1	Trunk, DID, Extension	1	Blue/White	26	White/Blue
2	Trunk, DID, Extension	2	Orange/White	27	White/Orange
3	Trunk, DID, Extension	3	Green/White	28	White/Green
4	Trunk, DID, Extension	4	Brown/White	29	White/Brown
5	Trunk, DID, Extension	5	Slate/White	30	White/Slate
6	Trunk, DID, Extension	6	Blue/Red	31	Red/Blue
7	Trunk, DID, Extension	7	Orange/Red	32	Red/Orange
8	Trunk, DID, Extension	8	Green/Red	33	Red/Green
9	Extension	9	Brown/Red	34	Red/Brown
10	Extension	10	Slate/Red	35	Red/Slate
11	Extension	11	Blue/Black	36	Black/Blue
12	Extension	12	Orange/Black	37	Black/Orange
13	Extension	13	Green/Black	38	Black/Green
14	Extension	14	Brown/Black	39	Black/Brown
15	Extension	15	Slate/Black	40	Black/Slate
16	Extension	16	Blue/Yellow	41	Yellow/Blue
17	Extension	17	Orange/Yellow	42	Yellow/Orange
18	Extension	18	Green/Yellow	43	Yellow/Green
19	Extension	19	Brown/Yellow	44	Yellow/Brown
20	Extension	20	Slate/Yellow	45	Yellow/Slate

Table D-14 IPBX-24 RJ-21X Telephone and Trunk Connector Pins

Port	Type	Ring		Tip	
		Pin #	Cable Color	Pin #	Cable Color
21	Extension	21	Blue/Violet	46	Violet/Blue
22	Extension	22	Orange/Violet	47	Violet/Orange
23	Extension	23	Green/Violet	48	Violet/Green
24	Extension	24	Brown/Violet	49	Violet/Brown
–		25	Slate/Violet	50	Violet/Slate

Table D-14 IPBX-24 RJ-21X Telephone and Trunk Connector Pins

D.5.2 IPBX-T1 and IPBX-E1 Voice Switch

The IPBX-T1 (Figure D-15) provides higher-density trunking to the central office using T1 or PRI signaling. The IPBX-E1 provides higher-density trunking using E1 PRI signaling.

The IPBX-T1 and IPBX-E1 LEDs and connectors are defined below.

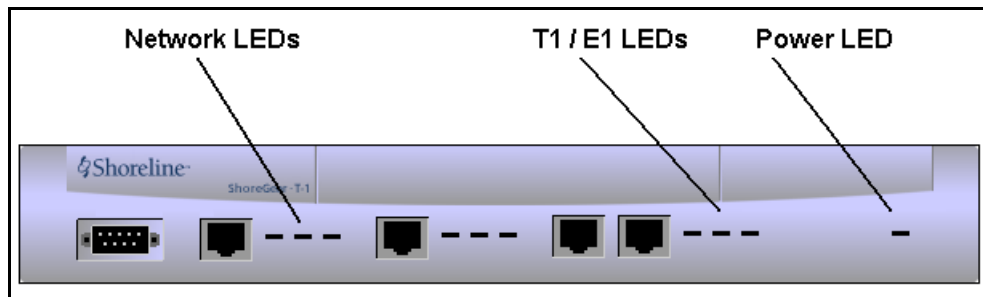


Table D-15 ShoreGear T1 and ShoreGear E1 Front Panel

D.5.2.1 LED Descriptions

Power LED

The IPBX-T1 and IPBX-E1 have one red power LED, which indicates the following:

- On: The switch is operating normally.
- Off: The switch has no power.
- Flashing
 - *2 flashes*—The switch failed its internal self-test. This indicates a hardware failure; replace the unit and submit a Return Material Authorization (RMA) to ShoreTel, Inc.
 - *3 flashes*—Booting via FTP. Flash memory may be corrupted. Go to the **Quick Look** page to make sure that the system is running properly.
 - *4 flashes*—The IP address is unavailable. DHCP and BOOTP did not respond to the IP address request, and the IP address is not available in nonvolatile memory to continue boot process. The switch will automatically reboot in five seconds and try again. Check the BOOTP/DHCP server and the network configuration to ensure that the voice switch is receiving a valid IP address.
 - *5 flashes*—The operating system is not available. The switch is booting from FTP but cannot find the boot files. It automatically reboots in five seconds.

- **6 flashes**—Using a previously stored IP address. A BOOTP/DHCP transaction was attempted, but the BOOTP/DHCP server did not respond. The switch continues to use the IP address stored in nonvolatile memory until it receives a valid response. If the switch receives a response that provides a different IP address, it reboots using the new IP address. If the switch receives a response that matches the IP address stored in nonvolatile memory, it continues operation, and the power LED stops flashing. If the problem persists, check the BOOTP/DHCP server and network configuration.

Network LEDs

Network LEDs indicate the speed at which the switch is communicating with the network and when there is network activity. The network LED descriptions follow:

- **Link:** This LED indicates that the switch is connected to an Ethernet network.
- **100M**
 - *When lit*, the switch is connected to a 100BaseT network.
 - *When off*, the switch is connected to a 10BaseT network.
- **Traffic:** This LED indicates network activity, as follows:
 - When flashing, network activity is detected.
 - When on (not flashing), heavy network activity is detected.
 - When off, network activity is not detected.

This LED is not directly related to any particular switch's individual network activity. For example, if you have three switches connected to the same hub, and one switch's Traffic LED shows activity, the other switches will indicate the same.

T1 and E1 LEDs

The T1 and E1 LEDs indicate line coding, network framing, and loopback status. These LEDs are color coded—green, yellow, and red. T1 and E1 LED alarms and errors are logged as switch events in ShoreWare Director's event log.

- **Line Coding:** This LED indicates line coding status, as follows:
 - *When green*, the line coding signal is good.
 - *When yellow*, bipolar violations (BPV) are detected one second intervals.
 - *When red*, a loss of signal (LOS) has occurred.
 - *When off*, the switch has no power.
- **Framing:** This LED indicates network framing status, as follows:
 - *When green*, the T1/E1 signal is *in frame*; the signal is synchronized.
 - *When yellow*, a *yellow alarm* has been received from the CO.
 - *When flashing yellow*, the frame-bit error rate has exceeded its limits.
 - *When flashing slow yellow*, frame-bit errors (>1 per million) have occurred; this state will take up to 10 minutes to clear.
 - *When flashing fast yellow*, frame-bit errors (>1 per 1000) have occurred.
 - *When red*, the T1/E1 signal is out-of-frame (OOF). The received signal cannot be framed to the Extended Superframe (ESF) or D4 format.
 - *When off*, the switch has no power.
- **Loopback:** This LED indicates loopback status for testing, as follows:
 - *When yellow*, loopback is enabled. This may be set from within ShoreWare Director or from the CO.
 - *When off*, loopback is disabled.

D.5.2.2 IPBX-T1 and IPBX-E1 Connectors

The IPBX-T1 and IPBX-E1 voice switches contain the following components:

- 1 DB-9 female connector for maintenance
- 1 RJ-45 connector for the LAN interface
- 1 RJ-48C connector for T1 or E1 monitoring
- 1 RJ-48C connector for the T1 or E1 interface

Symbols

# of Extension Digits parameter	
described	23

Numerics

0+ calling	100
1010nnn	99
311 calling	178, 179, 182
411 calling	99, 178, 179, 182
611 calling	99
800 calling	99
900 calls	178, 182
911 calling	99

A

Accept Broadcast Messages parameter	195
Access codes	178, 179, 182
Account Code Collection parameter	186
Account Codes	186
Adding individual loop-start trunks	109, 110
Allow Access to Broadcast Distribution List field	184
Allow Access to System Distribution Lists	184
Allow Call Pickup field	176
Allow Extension Reassignment parameter	176
Allow External Call-forwarding field	178
Allow Message Notification to External Number field	185
Allow PSTN Failover parameter	176
Allow Trunk-to-Trunk Transfer field	176
AMIS	275
Broadcast DL	277
disabling systems	279
enabling	276
maximum message length	276
restrictions	276
setup and tear-down	275
System DL	277
test mailbox	279
AMIS Voice Mail	275
analog ports	194
Area Codes	41
area codes	182
Auto-Attendant	
configuring schedules	293
Auto-attendant	
described	285
Auto-attendant parameters	
Custom	286, 305
DID	286

Extension	286
Holiday	286, 305
Invalid Entry	289
Menu Name	286
Multiple-Digit	289
On-Hours	286, 304
Operation	288
Prompt Text	288
Schedule	288
Time Out	289
Too Many Errors	289
auto-attendant	99
avoiding overlap of extensions	276

B

bandwidth management	112
Barge In	301
Barge In field	177
BCA <i>see bridged call appearance</i>	
boot parameters, IP phones	111
bridged call appearance, description	137
Broadcast Distribution Lists	184
Broadcast messages	195

C

call agent switch, hosting IP phone	112
Call Control Options edit page	176
Call Control Options parameters	
Admission control algorithm assumes RTP header compression is being used	155
Enable Monitor / Record Warning Tone	154
Generate an event when a trunk is in use for X minutes	154
Hang Up Make Me Conference after X minutes of silence	154
Park Timeout after nnnnnn seconds	154
Use Distributed Routing Service	154
Voice Encoding and Quality of Service	155
Call Control page	133
Call handling mode parameters	
Always Destination	144, 152, 235
Call Manager	196
login name	198
password	198
Call parameter	185
Call permissions	185
all	182
all calls	178, 182

- internal only 182
- international long distance 178
- local only 178
- national long distance 178
- Call permissions parameters 181
 - International Long Distance 182
 - Local Only 182
 - Name 181
 - National Long Distance 182
 - Permissions 182
 - Restrictions 182
 - Scope 181
- Caller ID blocking 100
- carrier codes 99
- Carrier Select numbers 178, 182
- CESID 41, 189
- CESID, caller id emergency services 113
- changeme default password 198
- Class of Service 185, 186
- Class of service
 - system distribution lists 273
- Client Password parameter 198
- Client Type Parameter
 - Personal 196
- Client Type parameter 188, 196
 - Operator 196
 - Workgroup Agent 196
 - Workgroup Supervisor 196
- Client User ID parameter 198
- Collaborative Servers parameters
 - Fully Qualified Domain Name 271
 - Name 271
 - Override Default Port 271
 - Site 271
 - Use HTTP 271
- Conference Bridge Server 271
- conference call handling 176
- Conference Port Status, checking 272
- conference server 271
- conferencing, setting up 271
- Configuration pages
 - described 8
- Control buttons
 - Copy 11
 - Delete 11
 - New 10
 - Reset 11
 - Save 11
- Converged Conference Bridge, and IP phone ports 111
- cookie use 192, 193
- COS Call Permissions parameter 186
- COS Telephony parameter 186
- COS Voice Mail parameter 186
- country code 178, 182
- Current Port parameter 194

D

- Data-entry fields
 - conventions ii

- Data-entry frame
 - described 8
- Default password 198
- default password 198
- Dial by Name 195
- DID 98, 188, 306
 - trunk group 194
- Digit collection 98
- digit collection 114
- Digit Translation Table 19
- digital tie trunk support 98
- Directed Intercom/Paging field 177
- Director
 - login name 198
 - password 198
- Disabled account code collection 186
- Distributed Routing Service 152
- DNIS 306
- DNIS 98

E

- Edit DID Range link 98
- Edit DNIS Map link 98
- E-mail address 198
- Emergency Call Back Number 41
- Enable Fax Redirection parameter 195
- enable international calls 99
- enable local calls 99
- enable long-distance calls 99
- Enable Message Notification parameter 309
- Event categories
 - Java Client 362
 - Java Server 362
 - Notification 362
 - Port Mapper 363
 - Switch 363
 - System Management Database 363
 - System Management Interface 363
 - Voice Mail Application 363
 - Voice Mail Message Server 363
 - Voice Mail Port Mapper 363
- Event filters
 - editing 367
- Event filters parameters
 - Category 367
 - Event ID 367
 - Source 367
 - Target E-mail Address 368
 - Type 367
- Event log 362
 - errors 358
- Executives user group 194
- Extension parameter 98, 188

F

- Fax redirection 195
- Find Me Destinations 178
- First System Dist. List Number parameter

described 17
 Forced account code collection 186
 forward a message 183

G

General information parameter
 Edit System Directory Record 199
 General information parameters
 Accept Broadcast Messages 195
 Allow Use of Soft Phone 196
 Client Password 198
 Client Type 196
 Client User ID 198
 Current Port 194
 DID 193
 Email Address 198
 email Address 198
 Enable Fax Redirection 195
 Home Port 194
 Include in Dial By Name Directory 195
 Jack # 194
 Mailbox 195
 Make Number Private 195
 Name 192
 Number 192
 Site 194
 Switch Port 194
 TUI Password 198
 User Group 194
 Voice Mail Password 198
 Go Home command 194

H

Handsfree Mode parameter 200
 Holiday Schedule, configuring 296
 Home Port parameter 194
 HTTP port use 271
 HTTPS (default port 443) 271
 Hunt Groups
 configuring schedules 293
 Hypertext links
 conventions ii

I

Inbound settings parameter 98
 Include in Dial by Name Directory parameter 195
 Incoming Max Messages field 183
 Individual loop-start trunk parameters 109, 110
 Individual users
 Client Type 188
 Extension 188
 Mailbox 188
 Name 187
 Port # 188
 Site 187
 Status 188
 Switch 188

 system directory 315
 User Group 188
 Internal Only call permissions 182
 International Long Distance call permission 178, 182
 IP Address Map 112
 IP address range 112
 IP address range, editing for a site 112
 IP Address Ranges, setting 111
 IP phone
 adding users 115
 announcement 114
 enabling failover 114
 how to move 119
 password 114
 setting options 113
 IP phones
 assigning by user 116
 renaming 118
 viewing 117

J

Jack # parameter 194

L

legacy PBXs 98
 legacy voice mail 19
 Licenses 196
 list pages
 described 8
 Local Only call permission 178, 182
 Login
 password 5
 login name 198
 Loop-start group parameters
 Name 96

M

Mailbox parameter 188, 195
 Maintenance
 event categories 362
 quick look services 357
 Make Me conference 176
 Make Me conference, six participants 272
 Make Me conferencing 272
 Make Number Private parameter 195
 Max Call Stack Depth field 176
 Max Parties in Make Me Conference parameter 176
 Max. TUI Errors parameter
 described 23, 136
 Menu Name parameter 286
 message notification 185
 messaging multiple users 273
 modes, starting and stopping 296
 Music-on-hold
 audio input port 485, 488, 491, 510, 513, 517
 mysql
 troubleshooting 418

MySQL database 417–425

N

Name (Call Permissions) field 181
 Name (Individual User) parameter 192
 Name (Individual Users) parameter 187
 Name (User Group) field 185
 Name (User Group) parameter 186
 Name (Users) field 175
 Name (voice mail) field 183
 National Long Distance call permission 178, 182
 National Mobile 178
 navigation frame
 described 6
 Night bell 41
 night bell 176
 Nightbell
 audio output port 484, 487, 491, 510, 513, 517, 529
 Notify New Users 228
 Notify New Users parameters
 Method 228, 230
 Status 228
 User Name 228, 230
 Number parameter 192

O

off-system mailboxes 276
 On-Hours Schedule, configuring 294
 Operator 196
 Operator Assisted calls 178, 182
 Optional account code collection 186
 Other site parameters
 # of Extension Digits 16
 First System Dist. List Number 17
 Last System Dist. List Number 17
 Max. GUI Password Length 24
 Max. TUI Errors 23, 136
 Min. GUI Password Length 24
 Outgoing Message Length field 183
 Outgoing Trunk Groups parameter 187
 Override Default Port with NNNN 271

P

Paging system
 audio output port 485, 488, 491, 510, 513, 517
 parked calls 176
 Password
 login 5
 password, IP phones 114
 patch-panel jack number 194
 Permissions field 179, 182
 personal assistant 42
 Personal client type 196
 Personal Options page 200
 Personal options parameters 200
 Call Waiting Tone Enabled 200
 Current call stack size 200

Default Trunk Group 200
 Edit Call Handling Modes 201
 Edit System Directory Record 199
 Port parameter 188
 PRI configuration 98
 Private numbers 195
 PSTN Failover 194
 PSTN numbers 176

Q

queued messages 183
 Quick Look
 system maintenance 346
 Quick look
 events 357, 358
 refresh 346, 384
 Switches 346
 Quick look event log
 Errors 358
 Informational 360
 Warnings 358

R

reassign an extension 176
 resetting IP phones 119
 Restrictions field 178

S

schedule
 configuring for holidays 296
 format for end time 298
 schedules
 configuring custom 297
 custom 293
 determining which is active 293
 Holiday 293
 Off-Hours 294
 On-Hours 293, 294
 schedules, configuring 293
 Scope field 181
 Server parameters
 Name 48
 NameAssigned User Group 50
 NameAuto-Attendant Extension 50
 NameCOM Port 51
 NameDefault Auto-Attendant Menu 51
 NameExtension List 51
 NameHost IP Address 48
 NameMessage Desk Number 51
 NameNumber of Digits 51
 NameSite 49
 NameSoftSwitch Name 49
 NameUse for Call Data 51, 52
 NameUse for MWI Data 51, 52
 NameVoice Mail Extension 50
 NameVoice Mail Login Extension 50
 ShoreGear 220E1 switch

- telephone ports 488
 - ShoreGear 220T1 switch
 - telephone ports 482, 491, 494
 - ShoreGear 220T1A switch
 - telephone ports 485
 - ShoreGear 50 switch
 - telephone ports 469, 472, 479, 501
 - ShoreGear 90 switch
 - telephone ports 466, 475, 498
 - Shoreline as PBX 19
 - ShorePhone-IP110, resetting 119
 - ShoreWare Director administration and maintenance client
 - control buttons 10
 - data-entry frame 8
 - list pages 8
 - login page 5
 - Show Call Manager Users parameter 187
 - site
 - editing IP address range 112
 - High IP Address 112
 - Low IP Address 112
 - Site page 178, 182
 - Site parameter 187, 194
 - Sites parameters
 - Additional Local Area Codes 41
 - Admission Control Bandwidth 42
 - Country 40
 - Edit IP Phone Address Map 42
 - Emergency Call Back Number 41
 - Emergency Number 42
 - Fax Redirect Extension 42
 - Local Area Code 41
 - Name 40
 - Night Bell Switch 41
 - Nightbell Extension 41
 - Operator Extension 41
 - Paging Extension 41
 - Paging Switch 41
 - Parent 40
 - Time Zone 41
 - Use Parent As Proxy 41
 - SoftPhone
 - User general information parameters
 - Allow Use of Soft Phone 196
 - stacked calls 176
 - Status parameter 188
 - Switch parameter 188
 - switch port 194
 - switch ports 194
 - System Directory 199
 - System directory
 - Go To This User Link 317
 - Name 316
 - outside contacts 315
 - System directory data-entry fields
 - Cell Phone 317
 - E-mail Address 317
 - Fax Phone 317
 - First Name 316
 - Home Phone 317
 - Last Name 316
 - Pager 317
 - Work Phone 317
 - System Directory Entry page 199
 - System Distribution List
 - configuring parameters 274
 - system distribution list
 - editing 274
 - System distribution list parameters 274
 - AMIS Systems 275
 - Name 274
 - Number 274
 - Type or select from list 275
 - System Distribution Lists 184
 - System distribution lists
 - Edit User page 273
 - System extensions parameters
 - Account Code Extension 21
 - Auto-Attendant Extension 21
 - Make Me Conference Extension 21
 - Voice Mail Broadcast Mailbox 21
 - Voice Mail Forward Extension 20
 - Voice Mail Login Extension 21
 - System maintenance
 - quick look 346
- ## T
- telephone port 188
 - Telephony Features parameter 185
 - Telephony features permissions parameters
 - Allow Call Pickup 176
 - Allow Call Transfer Trunk to Trunk 176
 - Allow Extension Reassignment 176
 - Allow External Call-forwarding & Find Me Dest .. 178
 - Allow PSTN Failover 176
 - Barge In 177
 - Directed Intercom/Paging 177
 - Max Parties in Make Me Conference 176
 - Max. Call Stack Depth 176
 - Permissions 179
 - Restrictions 178
 - teleworking IP phones, adding 112
 - Time Zone parameter 41
 - toll fraud 176
 - Trunk Group parameters
 - 911 99
 - Called ID not blocked by default 100
 - Destination 99
 - DID 98
 - DNIS 98
 - Easy Recognizable Codes 99
 - Explicit Carrier Selection 99
 - Extension 98
 - Inbound 98
 - International calls 99
 - Local calls 99
 - Long Distance calls 99
 - n11 calls 99
 - Operator Assisted 100

Outbound-Trunk Services	99
Trunk Digit Manipulation	100
trunk ports	466, 469, 472, 475, 479, 482, 485, 488, 491, 494, 498, 501
trunk-to-trunk transfer	176

U

User general information parameters	
Accept Broadcast Messages	195
Client Password	198
Client Type	196
Client User ID	198
Current Port	194
Edit System Directory Record	199
Email Address	198
email Address	198
Enable Fax Redirection	195
Home Port	194
Include in Dial By Name Directory	195
Jack #	194
Mailbox	195
Make Number Private	195
Name	192
Number	192
PSTN Failover	194
Site	194
Switch Port	194
TUI Password	198
User Group	194
Voice Mail Password	198
User Group parameter	188, 194
User Groups parameters	
Name	185
Telephony Features	185
Voice Mail	185
User groups parameters	186
Outgoing Trunk Groups	187
Show Call Manager users a list of account codes when dialing	187
User groups paramters	
Account Code Collection	186
COS-Call Permissions	186
COS-Telephony	186
COS-Voice Mail	186
Name	186
User Groupsparameters	
Call	185
User personal options parameters	
Call Waiting Tone Enabled	200
Current Call Handling Mode	201
Current Call Stack Size	200
Current call stack size	200
Default Trunk Group	200
Edit Call Handling Modes	201
Edit System Directory Record	199
Handsfree Mode	200
Ring Type	200

V

Vertical Service Codes	100, 182
virus checkers	418
voice codec, inter-site	112
Voice Mail	
password	198
setting options	277
setting User Group permissions	279
voice mail	
AMIS	275
Voice mail notification parameters	
Notification Number	204
Pager Data	204
Pager ID	204
Retry Count	205
Retry Interval	205
Voice Mail Notification Condition	204, 230
Voice Mail Notification Method	204
Voice Mail parameter	185
Voice mail permissions parameters	183
Allow Access to Broadcast Distribution List	184
Allow Access to System Distribution Lists	184
Allow Notification	185
Allow Notification to External Number	185
Incoming Max. Messages	183
Incoming Message Length	183
Name	183
Outgoing Message Length	183

W

Workgroup Agent	196
Workgroup Supervisor	196
Workgroups parameters	
Accept Broadcast Messages	307
Backup Extension	306
Call Forward	308
Call Handling Distribution Pattern	308
Current Call Handling Mode	308
DID	306
DNIS	306
Enable Automatic Agent Logout On Ring No Answer	307
Enable Message Notification	309
Extension	306
Include in Dial By Name Directory	307
Logged Out	309
Mailbox	306
Mailbox Workgroup Greeting	309
Name	306
Recorded Name	307
Rings per Agent	309
Schedule	308
User Group	306
Voice Mail Password	307
Workgroup Assistant	309
Workgroup Membership	307
Workgroup Queue Handling	307
Wrap Up Time	308